



Beijing Flying Voice Co., Ltd.

APX5008

User Manual

V1.0

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1 Introduction

1.1 Thank you for purchasing

Thank you for purchasing APX5008, APX5008 is a highly integrated, reliable, stable device, which includes rich Router and IP PBX functions. This manual introduces and describes how to install, how to configure and how to use APX5008, and it provides customers with a series of office configuration cases. User can use APX5008 to set your own IP Phone system. APX5008 provides a open source embedded IP PBX system, it runs ulinux and Asterisk and supports abundant IP PBX features.

1.2 Appearance and Installation

1.2.1 Appearance

APX5008 is a silver alloy device, user can see different ports on the panel of APX5008, you can see 1 reset console, 1 serial console, 4 LAN ports, 1 USB port, 1 SD card slot and 8 FXO ports on the front panel of APX5008. The following picture shows the ports and the table shows the functions of different ports:



| Port | Description |
|----------------|--|
| Reset | Press reset button 5s to make APX5008 factory default. |
| Serial Console | Connect it to the com port of PC to enter APX5008's Linux interface. |
| Internet | Connect to Internet or router. |
| LAN | Connect to PC or build small LAN network. |
| USB | Access for USB storage device. |
| SD | Used to insert the SD memory card. |
| FXO | Used to connect to the analog line. |

There is a power port and a power button on the back panel of APX5008.



1.2.2 Package Contents

- ◆ One APX5008
- ◆ Two L-shape brackets and some screws
- ◆ One serial console line
- ◆ One 4G SD card
- ◆ One power line

1.2.3 Installation

If user wants to use APX5008 as a gateway, please insert one end of the Ethernet cable into APX5008's Internet port and insert the other end of the Ethernet cable to your existing broadband connection port. And then log in APX5008's webpage to configure its Internet access ways, such as Static IP, DHCP or PPPoE;

If user wants to use APX5008 as bypass access, you don't need to connect to the Internet port; Insert one end of Ethernet cable into one of the LAN ports, and insert the other end of the Ethernet cable to your local network. Then connect the power adapter to the power port on the back panel of APX5008, and plug another end of power port into a wall outlet or power strip. And then please turn on APX5008.

The default IP for APX5008's LAN port is 192.168.0.1. Put the default IP in your web browser and it will redirect to the setting page of APX5008, the default username and password for the web access is:

Username: root

Password: admin

If you can't access APX5008, please check if you have connected the RJ-45 cable to the LAN port and if your computer is in the same network 192.168.0.xxx as APX5008.

Notice: the recommended web browser for logging in APX5008 is Firefox.

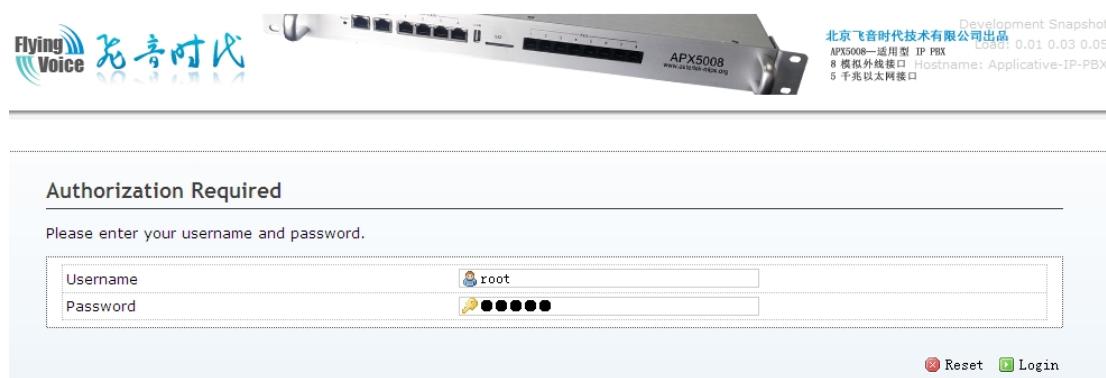
2 Configure the Router

2.1 Basic Configuration

2.1.1 System Status

Access APX5008's router configuration page.

The default IP address of APX5008's LAN port is 192.168.0.1, the username is 'root', password is 'admin', the login page is as the following picture.



After enter the password, you will log in in the router status page of APX5008.



System

- ◆ Router Name: display the router name;
- ◆ Router Model: display the router model;
- ◆ Firmware Version: display the current firmware version;

- ◆ Kernel Version: display the kernel version;
- ◆ Local Time: display the current time;
- ◆ Uptime: display the uptime since user turn on APX5008;
- ◆ Load Average: display the average load, the first figure represents the average load in 1 minute, and the second figure represents the average load in 5 minutes and the last one represents the average load in 15 minutes;

Memory

Display the current usage status of the memory.

Network

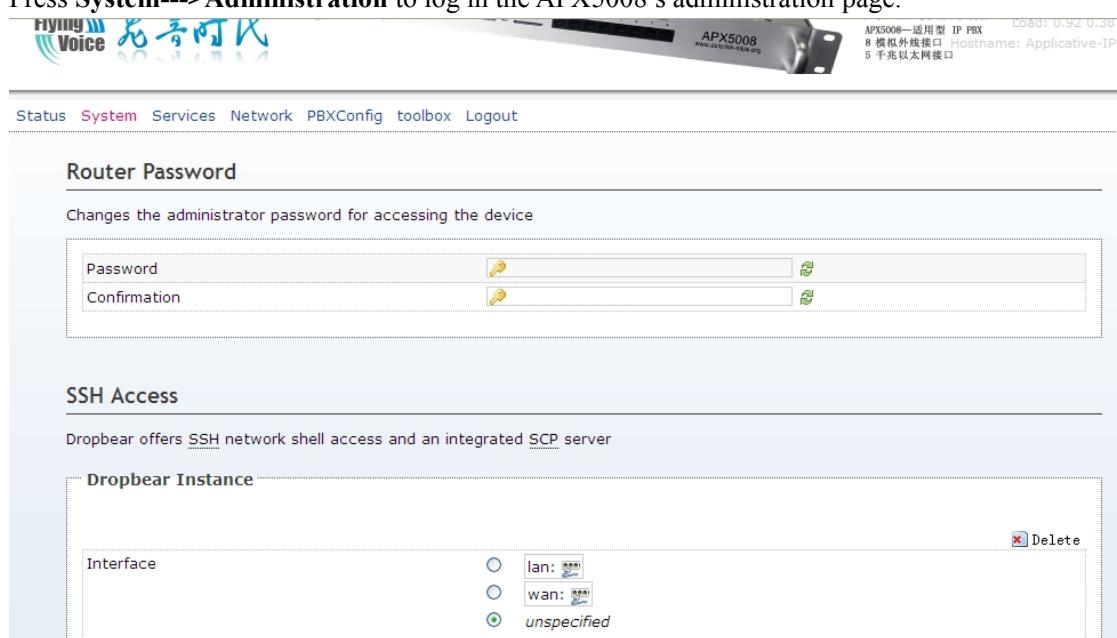
Display the current connection status of the Internet ports, including IP mode, IP address, gateway, connected time and the active connections.

DHCP Leases

Display the current IP addresses which APX5008 assigns to the DHCP client.

2.1.2 Set the Login Password

Press System--->Administration to log in the APX5008's administration page.



The screenshot shows the APX5008 Administration interface. At the top, there is a banner with the device model "APX5008", its function "适用型 IP PBX", and its load status "Load: 0.92 0.38". Below the banner, the navigation bar includes links for Status, System, Services, Network, PBXConfig, toolbox, and Logout. The main content area has two sections:

- Router Password**: A form for changing the administrator password, containing fields for "Password" and "Confirmation" with key icons.
- SSH Access**: A section for Dropbear Instance configuration, showing three radio button options for "Interface": "lan:" (selected), "wan:", and "unspecified". There is also a "Delete" button.

Router Password

User can set or change the router password. Click the Save&Reply button which is at the lower right corner of the page after you finish entering.

SSH Access

Set the parameters of SSH access, such as interface, port, password authentication and so on.

2.1.3 Configure Network Interfaces

Click Network-->Interface, then user can login the network interface configuration page of APX5008, just like the following picture:

Interfaces

Interface Overview

| Network | Status | Actions |
|--|---|--|
| LAN  br-lan | Uptime: 0h 5m 27s MAC Address: 00:21:22:11:BB:AA RX: 108.38 KB (1837 Pkts.) TX: 386.02 KB (509 Pkts.) IPv4: 192.168.30.1/24 IPv6: FE80:0:0:0:221:22FF:FE11:BBAA/64 |  Connect  Stop  Edit  Delete |
| WAN  eth0.2 | Uptime: 0h 0m 0s MAC Address: 00:21:22:11:BB:AA RX: 0.00 B (0 Pkts.) TX: 44.21 KB (116 Pkts.) IPv6: FE80:0:0:0:221:22FF:FE11:BBAA/64 |  Connect  Stop  Edit  Delete |

 Add new interface...

Both LAN port and WAN port have four options: Connect, Stop, Edit and Delete, functions of each options are as following:

- ◆ Connect: Click “Connect”, APX5008 should disconnect the current network and connect again;
- ◆ Stop: Click “Stop”, APX5008 should stop the current network connections;
- ◆ Edit: Click “Edit”, user can configure the corresponding network interfaces;
- ◆ Delete: Click “Delete” to remove the current interface.(Please take care of this option)

2.1.3.1 Configure WAN port

Click “Edit” option after WAN port, then user can begin to configure WAN port.

Common Configuration

| General Setup | | Advanced Settings | Physical Settings | Firewall Settings |
|--|-------------------------------------|---|-------------------|-------------------|
| Status Uptime: 0h 0m 0s MAC Address: 00:21:22:11:BB:AA RX: 0.00 B (0 Pkts.) TX: 595.92 KB (1485 Pkts.) IPv6: FE80:0:0:0:221:22FF:FE11:BBAA/64 | | | | |
| Protocol | Static address |  | | |
| IPv4 address | <input type="text"/> | | | |
| IPv4 netmask | <input type="text"/> |  | | |
| IPv4 gateway | <input type="text"/> | | | |
| IPv4 broadcast | <input type="text"/> | | | |
| Use custom DNS servers | <input type="text"/> |  | | |
| Accept router advertisements | <input type="checkbox"/> | | | |
| Send router solicitations | <input checked="" type="checkbox"/> | | | |
| IPv6 address | <input type="text"/> | | | |
| IPv6 gateway | <input type="text"/> | | | |

IP Aliases
This section contains no values yet



Firstly, user can configure the basic settings in General Setup column.

Status: Check the current information of WAN port .

- ◆ Protocol: Choose the connection type of WAN port, options are Static address, DHCP client, PPPoE and so on, if user wants to change the connection type, please click  , then user can begin to configure the corresponding parameters.

◆ DHCP Client

Hostname to send when requesting DHCP: the IP address of DHCP server that user assigns.

◆ Static Address:

IPv4 address: IPv4 address

IPv4 netmask: IPv4 netmask

IPv4 gateway: IPv4 gateway

IPv4 broadcast: IPv4 broadcast

Use custom DNS servers: enter the IP address of the custom DNS server

◆ PPPoE

PAP/CHAP username: the username of your PPPoE account

PAP/CHAP password: the password of your PPPoE account

Access Concentrator: the IP address of access concentrator, leave empty to autodetect

Service Name: the address of PPPoE server, leave empty to auto detect.

In **Advanced Settings** column, user can configure some other parameters, such as MAC, MTU and so on. (*Notice: when user choose different IP protocol, the parameters in this column should be different.*)

◆ IP-Aliases: This option is used to add another IP address for this interface, in order to make one network interface correspond to multiple IP. Please enter an alias in the IP_alias text box, such as "flyingvoice", and then click  , then user can configure the corresponding parameters, including IPv4-Address, IPv4-Netmask, IPv4-Gateway, IPv4-Broadcast and DNS-Server, the following picture shows the details of the parameters:

| IP-Aliases | |
|---|-------------------|
| FLYINGVOICE | |
| General Setup | Advanced Settings |
| IPv4-Address | 192.168.40.1 |
| IPv4-Netmask | 255.255.255.0 |
| IPv4-Gateway | 192.168.20.1 |
|  | |

Click Delete to remove current IP-Aliases.

2.1.3.2 Configure LAN port

It is the same to configure the IP protocol of LAN port with WAN port, what is more, user can use LAN port to set as DHCP server, just like the following picture:

| DHCP Server | |
|------------------|--|
| General Setup | Advanced Settings |
| Ignore interface | <input type="checkbox"/>  Disable DHCP for this interface. |
| Start | 100  Lowest leased address as offset from the network address. |
| Limit | 150  Maximum number of leased addresses. |
| Leasetime | 12h  Expiry time of leased addresses, minimum is 2 Minutes (2m). |

- ◆ Ignore interface: check this option, the DHCP server of this interface should be disabled.
- ◆ Start: the starting IP address the DHCP server assigns
- ◆ Limit: how many IP addresses should this server assigns
- ◆ Lease time: set expiry time of leased addresses, minimum is 2 Minutes(2m), and default is 12 hours.

2.2 More Functions and Configurations

2.2.1 Status Checking

2.2.1.1 Firewall Status

Click **Status-->Firewall**, user can check the status of firewall, reset counters and restart firewall.

Firewall Status

| Actions | | | | | | | | | | |
|---|-------|-----------|---------------------------|-------|-------|----|-----|-----------|-------------|-----------------------------|
| Table: Filter | | | | | | | | | | |
| Chain INPUT (Policy: ACCEPT, Packets: 0, Traffic: 0.00 B) | | | | | | | | | | |
| Rule # | Pkts. | Traffic | Target | Prot. | Flags | In | Out | Source | Destination | Options |
| 1 | 12884 | 1.76 MB | ACCEPT | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | ctstate RELATED,ESTABLISHED |
| 2 | 0 | 0.00 B | DROP | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | ctstate INVALID |
| 3 | 18 | 1.13 KB | ACCEPT | all | -- | lo | * | 0.0.0.0/0 | 0.0.0.0/0 | - |
| 4 | 2657 | 134.93 KB | syn_flood | tcp | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | tcp flags:0x17/0x02 |
| 5 | 2834 | 168.02 KB | input_rule | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | - |
| 6 | 2834 | 168.02 KB | input | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | - |
| Chain FORWARD (Policy: ACCEPT, Packets: 0, Traffic: 0.00 B) | | | | | | | | | | |
| Rule # | Pkts. | Traffic | Target | Prot. | Flags | In | Out | Source | Destination | Options |
| 1 | 0 | 0.00 B | ACCEPT | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | ctstate RELATED,ESTABLISHED |
| 2 | 0 | 0.00 B | DROP | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | ctstate INVALID |
| 3 | 0 | 0.00 B | forwarding_rule | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | - |
| 4 | 0 | 0.00 B | forward | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | - |
| Chain OUTPUT (Policy: ACCEPT, Packets: 0, Traffic: 0.00 B) | | | | | | | | | | |
| Rule # | Pkts. | Traffic | Target | Prot. | Flags | In | Out | Source | Destination | Options |
| 1 | 17313 | 6.43 MB | ACCEPT | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | ctstate RELATED,ESTABLISHED |
| 2 | 0 | 0.00 B | DROP | all | -- | * | * | 0.0.0.0/0 | 0.0.0.0/0 | ctstate INVALID |

2.2.1.2 Router Status

Click **Status--> Router**, user can check the current ARP table and router table:

Routes

The following rules are currently active on this system.

| ARP | | | |
|----------------|-------------------|-----------|--|
| IPv4-Address | MAC-Address | Interface | |
| 192.168.30.134 | 00:1c:25:70:ac:a7 | br-lan | |

| Active IPv4-Routes | | | |
|--------------------|-----------------|--------------|--------|
| Network | Target | IPv4-Gateway | Metric |
| lan | 192.168.30.0/24 | 0.0.0.0 | 0 |

| Active IPv6-Routes | | | |
|--------------------|--------------------|-----------------|------------|
| Network | Target | IPv6-Gateway | Metric |
| loopback | 0:0:0:0:0:0:0:0 | 0:0:0:0:0:0:0:0 | FFFFFFFFFF |
| loopback | 0:0:0:0:0:0:0:1 | 0:0:0:0:0:0:0:0 | 00000000 |
| lan | FF02:0:0:0:0:0:1:2 | 0:0:0:0:0:0:0:0 | 00000000 |
| (eth0) | FF00:0:0:0:0:0:0:8 | 0:0:0:0:0:0:0:0 | 00000100 |
| lan | FF00:0:0:0:0:0:0:8 | 0:0:0:0:0:0:0:0 | 00000100 |
| wan | FF00:0:0:0:0:0:0:8 | 0:0:0:0:0:0:0:0 | 00000100 |
| loopback | 0:0:0:0:0:0:0:0 | 0:0:0:0:0:0:0:0 | FFFFFFFFFF |

2.2.1.3 System Log

Click **Status --> System Log** to check APX5008's system log.

System Log

```

Jan 11 02:05:39 Applicative-IP-PBX kern.warn kernel: ^M*****bufsize:160*****numbufs:4*****
Jan 11 02:05:39 Applicative-IP-PBX kern.warn kernel: ^M*****bufsize:160*****numbufs:4*****
Jan 11 02:05:40 Applicative-IP-PBX daemon.notice openvpn(sample_server)[1579]: OpenVPN 2.x-testing-aa52ca828f
Jan 11 02:05:40 Applicative-IP-PBX daemon.notice openvpn(sample_client)[1581]: OpenVPN 2.x-testing-aa52ca828f
Jan 11 02:05:40 Applicative-IP-PBX daemon.warn openvpn(sample_client)[1581]: WARNING: No server certificate v
Jan 11 02:05:40 Applicative-IP-PBX daemon.warn openvpn(sample_client)[1581]: NOTE: OpenVPN 2.1 requires '--sc
Jan 11 02:05:40 Applicative-IP-PBX daemon.err openvpn(custom_config)[1577]: Options error: In [CMD-LINE]:1: E
Jan 11 02:05:40 Applicative-IP-PBX daemon.warn openvpn(sample_server)[1579]: NOTE: OpenVPN 2.1 requires '--sc
Jan 11 02:05:40 Applicative-IP-PBX daemon.err openvpn(sample_client)[1581]: Cannot load certificate file /etc
Jan 11 02:05:40 Applicative-IP-PBX daemon.warn openvpn(custom_config)[1577]: Use --help for more information.
Jan 11 02:05:40 Applicative-IP-PBX daemon.err openvpn(sample_server)[1579]: Cannot open /etc/openvpn/dh1024.p
Jan 11 02:05:40 Applicative-IP-PBX daemon.notice openvpn(sample_client)[1581]: Exiting due to fatal error
Jan 11 02:05:40 Applicative-IP-PBX daemon.notice openvpn(sample_server)[1579]: Exiting due to fatal error
Jan 11 02:05:40 Applicative-IP-PBX kern.warn kernel: ^M
Jan 11 02:05:40 Applicative-IP-PBX kern.cxit kernel: ramips-wdt: timeout value 60 must be 0 < timeout < 34
Jan 11 02:05:41 Applicative-IP-PBX user.info sysinit: astdatadir=/var/lib/asterisk^[[0;37mservice: file '/usr
Jan 11 02:05:41 Applicative-IP-PBX user.info sysinit: /etc/rc.common: line 82: s: not found
Jan 11 02:05:49 Applicative-IP-PBX kern.warn kernel: *****bufsize:160*****numbufs:4*****
Jan 11 02:05:49 Applicative-IP-PBX kern.warn kernel: ^M*****bufsize:160*****numbufs:4*****
Jan 11 02:06:21 Applicative-IP-PBX user.info sysinit: uci: Entry not found

```

2.2.1.4 Kernel Log

Click **Status --> Kernel Log**, user can check the kernel log.

Kernel Log

```

Linux version 2.6.39.4 (root@localhost.localdomain) (gcc version 4.5.4 20110808 (prerelease) (Linaro GCC 4.5-5)
prom: fw_arg0=00000001, fw_arg1=87f3fffb0, fw_arg2=07f403b0, fw_arg3=00000000
env[0]: memsize=128
env[1]: initrd_start=0x00000000
env[2]: initrd_size=0x0
env[3]: flash_start=0x00000000
env[4]: flash_size=0x1000000
env[0]: memsize=128
env[1]: initrd_start=0x00000000
env[2]: initrd_size=0x0
env[3]: flash_start=0x00000000
env[4]: flash_size=0x1000000
bootconsole [early0] enabled
CPU revision is: 0001974c (MIPS 74Kc)
Ralink RT3883 id:1 rev:5 running at 500.00 MHz
Determined physical RAM map:
    memory: 08000000 @ 00000000 (usable)
Initrd not found or empty - disabling initrd
Zone PFN ranges:
    Normal 0x00000000 -> 0x00008000
Movable zone start PFN for each node

```

2.2.1.5 Processes

Click **Status-->Processes**, user can check the processes of APX5008.

Processes

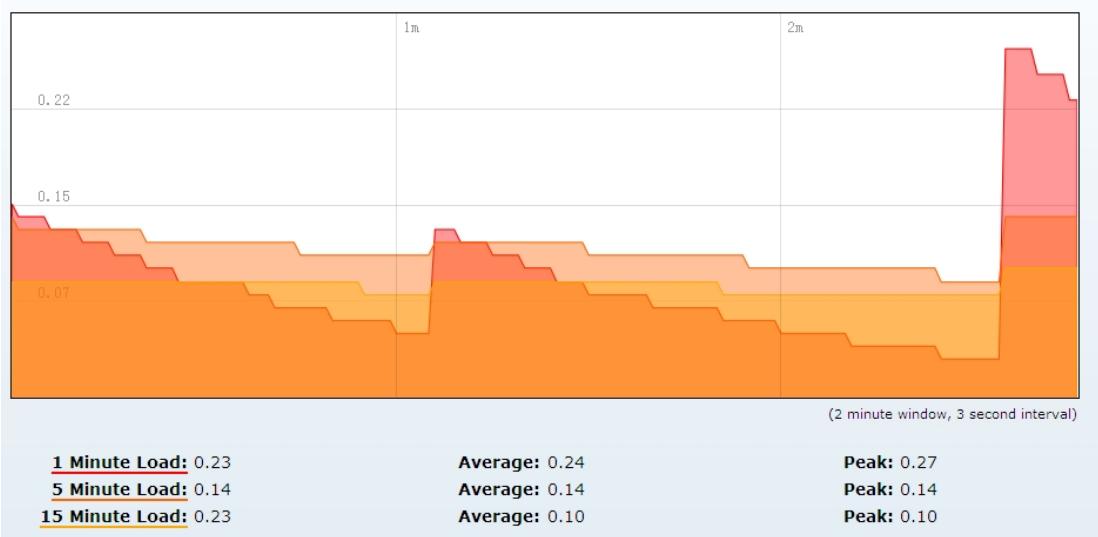
This list gives an overview over currently running system processes and their status.

| PID | Owner | Command | CPU usage (%) | Memory usage (%) | Hang Up | Terminate | Kill |
|-----|-------|---------------|---------------|------------------|---------|-----------|------|
| 1 | root | init | 0% | 1% | | | |
| 2 | root | [kthreadd] | 0% | 0% | | | |
| 3 | root | [ksoftirqd/0] | 0% | 0% | | | |
| 4 | root | [kworker/0:0] | 0% | 0% | | | |
| 5 | root | [kworker/u:0] | 0% | 0% | | | |
| 6 | root | [rcu_kthread] | 0% | 0% | | | |
| 7 | root | [khelper] | 0% | 0% | | | |
| 45 | root | [sync_supers] | 0% | 0% | | | |

2.2.1.6 Realtime Graphs

Click **Status--> Realtime Graphs**, user can view some realtime graphs, such as real time graphs of Load, Traffic, Wireless and Connections.

Realtime Load



2.2.2 System Configuration

2.2.2.1 System

Click **System--> System** to begin system configuration, in this webpage, user can configure the Hostname and Timetome, and refresh the local time for APX5008; what is more, user can configure the parameters of system log: such as system log buffer size, log level, log server and so on; lastly, user can change the Language and Style of APX5008, APX5008 support English and Chinese at this moment.

System

Here you can configure the basic aspects of your device like its hostname or the timezone.

| System Properties | |
|--|---------|
| General Settings | Logging |
| Language and Style | |
| Local Time | |
| Sat Mar 23 08:10:32 2013 <input checked="" type="checkbox"/> Sync with browser | |
| Hostname <input type="text" value="Applicative-IP-PEX"/> | |
| Timezone <input type="text" value="Asia/Shanghai"/> | |

2.2.2.2 Administration

Click **System--> Administration**, user can login the management webpage of APX5008, and configure the router login password and SSH Access.



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Router Password

Changes the administrator password for accessing the device

| | | |
|--------------|--------------------------|--|
| Password | <input type="password"/> | |
| Confirmation | <input type="password"/> | |

SSH Access

Dropbear offers SSH network shell access and an integrated SCP server

Dropbear Instance

| | | |
|-------------------------|--|--|
| Interface | <input type="radio"/> lan: <input type="radio"/> wan: <input checked="" type="radio"/> unspecified | |
| Port | 22 | Specifies the listening port of this Dropbear instance |
| Password authentication | <input checked="" type="checkbox"/> Allow SSH password authentication | |

2.2.2.3 Software

Click **System--> Software** to login the management webpage of system software, in this webpage, user can check the usage of Flash, install new software, remove the old firmware and so on.

Software

Actions Configuration

No package lists available Update lists
Free space: 89% (1.23 MB)

Download and install package:

Filter:

Status

Installed packages Available packages

| | Package name | Version |
|------------------------|--------------------------------|------------|
| Remove | asterisk-gui | svn-r5161 |
| Remove | asterisk18 | 1.8.10.1-4 |
| Remove | asterisk18-app-alarmreceiver | 1.8.10.1-4 |
| Remove | asterisk18-app-authenticate | 1.8.10.1-4 |
| Remove | asterisk18-app-chansavail | 1.8.10.1-4 |
| Remove | asterisk18-app-chanspy | 1.8.10.1-4 |
| Remove | asterisk18-app-directed-pickup | 1.8.10.1-4 |
| Remove | asterisk18-app-disa | 1.8.10.1-4 |

2.2.2.4 Startup

Click **System--> Startup** to login the startup management webpage, in this webpage, user can enable or disable the initscripts, and user can start/stop/restart the initscript.

Initscripts

You can enable or disable installed init scripts here. Changes will applied after a device reboot.

Warning: If you disable essential init scripts like "network", your device might become inaccesable!

| Start priority | Initscript | Enable/Disable | Start | Restart | Stop |
|----------------|--------------|--|-------|---------|------|
| 0 | umount | <input checked="" type="checkbox"/> Disabled | | | |
| 5 | luci_fixtime | <input checked="" type="checkbox"/> Enabled | | | |
| 10 | boot | <input checked="" type="checkbox"/> Enabled | | | |
| 127 | rcS | <input checked="" type="checkbox"/> Disabled | | | |
| 20 | fstab | <input checked="" type="checkbox"/> Enabled | | | |
| 39 | usb | <input checked="" type="checkbox"/> Enabled | | | |
| 40 | dahdi | <input checked="" type="checkbox"/> Enabled | | | |
| 40 | network | <input checked="" type="checkbox"/> Enabled | | | |
| 45 | firewall | <input checked="" type="checkbox"/> Enabled | | | |
| 50 | asterisk | <input checked="" type="checkbox"/> Enabled | | | |
| 50 | cron | <input checked="" type="checkbox"/> Enabled | | | |

2.2.2.5 Scheduled Tasks

Click **System-> Scheduled Tasks**, user can login the scheduled tasks page of APX5008, in this page, user can view the current tasks, and can add customized scheduled tasks.

Scheduled Tasks

This is the system crontab in which scheduled tasks can be defined.

```
*/5 * * * * killall -HUP dnsmasq
*/5 * * * * /usr/sbin/ff_olsr_watchdog
0 */4 * * * /usr/sbin/ff_rdate
```

2.2.2.6 Time Synchronization

Click **System-->Time Synchronization**, user can login the time synchronization page of APX5008, in this page, user can configure the update interval, offset frequency and time servers.

Time Synchronisation

Synchronizes the system time

General

| | |
|------------------------------|--|
| Current system time | Sat Mar 23 00:41:22 2013 |
| Update interval (in seconds) | 600 |
| Count of time measurements | <input type="text"/> <small>(? empty = infinite)</small> |

Clock Adjustment

| | |
|------------------|---|
| Offset frequency | 0 |
|------------------|---|

Time Servers

| Hostname | Port | |
|------------------------|------|--|
| 0.openwrt.pool.ntp.org | 123 | |
| 1.openwrt.pool.ntp.org | 123 | |
| 2.openwrt.pool.ntp.org | 123 | |
| 3.openwrt.pool.ntp.org | 123 | |
| | | |

2.2.2.7 Mount Points

Click **System-> Mount Points** to login the mount points page of APX5008, and in this page, user can check the current mounted file systems and mount points, what is more, user can enable SWAP or modify the parameters of current SWAP.

Mount Points

Mounted file systems

| Filesystem | Mount Point | Available | Used |
|--------------------|-------------|-----------------------|-----------------|
| rootfs | / | 1.23 MB / 1.39 MB | 11% (164.00 KB) |
| /dev/root | /rom | 0.00 B / 13.50 MB | 100% (13.50 MB) |
| tmpfs | /tmp | 61.15 MB / 62.19 MB | 2% (1.04 MB) |
| tmpfs | /dev | 512.00 KB / 512.00 KB | 0% (0.00 B) |
| /dev/mtdblock5 | /overlay | 1.23 MB / 1.39 MB | 11% (164.00 KB) |
| overlayfs:/overlay | / | 1.23 MB / 1.39 MB | 11% (164.00 KB) |
| /dev/sda1 | /sdcard | 3.32 GB / 3.60 GB | 3% (102.08 MB) |

Mount Points

Mount Points define at which point a memory device will be attached to the filesystem

| Enabled | Device | Mount Point | Filesystem | Options | Root | Check | | |
|-------------------------------------|---------------------|-------------|------------|---------|------|-------|--|--|
| <input checked="" type="checkbox"/> | /dev/sda1 (3748 MB) | /sdcard | ext4 | rw,sync | no | yes | | |

SWAP

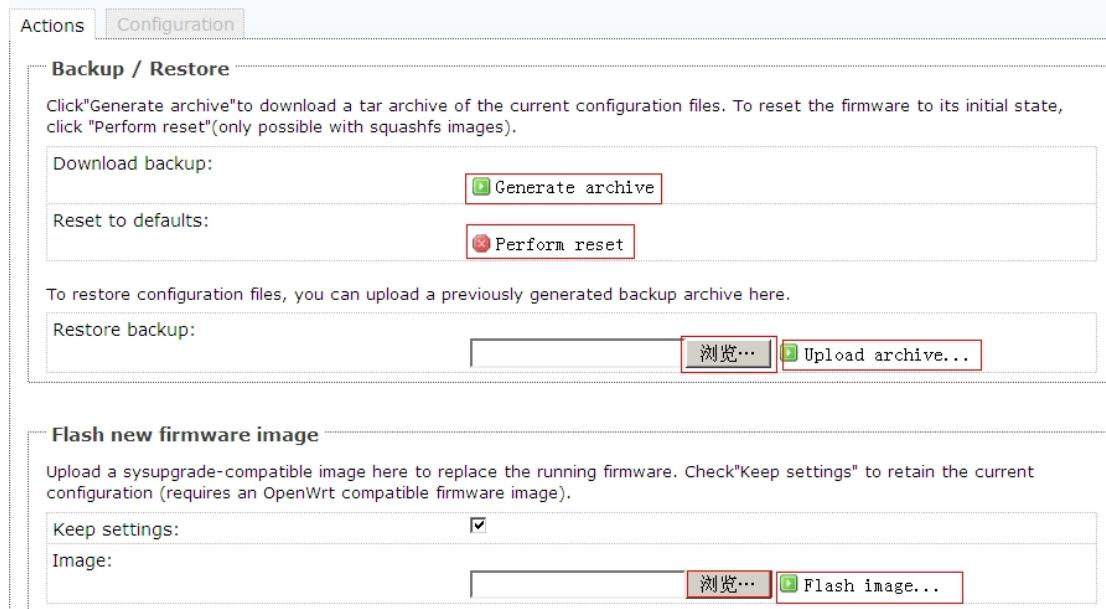
If your physical memory is insufficient unused data can be temporarily swapped to a swap-device resulting in a higher amount of usable RAM. Be aware that swapping data is a very slow process as the swap-device cannot be accessed with the high datarates of the RAM.

| Enabled | Device |
|--------------------------|-----------|
| <input type="checkbox"/> | /dev/sda2 |

2.2.2.8 Backup/Flash Firmware

Click **System--> Backup/Flash Firmware** to login APX5008's Flash Operations page, in this page, user can backup the configuration files of APX5008, and make APX5008 factory default, additionally, user can upgrade a new firmware for APX5008.

Flash operations



Actions Configuration

Backup / Restore

Click "Generate archive" to download a tar archive of the current configuration files. To reset the firmware to its initial state, click "Perform reset" (only possible with squashfs images).

Download backup:

Reset to defaults:

To restore configuration files, you can upload a previously generated backup archive here.

Restore backup:

Flash new firmware image

Upload a sysupgrade-compatible image here to replace the running firmware. Check "Keep settings" to retain the current configuration (requires an OpenWrt compatible firmware image).

Keep settings:

Image:

1) Download backup

Click , the browser will remind user to download the configuration file, click Save and select the directory, then the file will be saved in your local PC(the file contains all the configurations of APX5008)

2) Restore to defaults

Click to make APX5008 factory default.

Notice: A hard reset will remove all your settings, and your device will be restored to factory default settings. Do not perform a hard reset unless you have backed up your settings or you can do the configurations again.

3) Restore backup

This action will make user be able to restore all the configurations of APX5008.



Restore backup:

Click to select the file you need to restore and click

to upload the file. After you press button, the system will remind user that changes applied and system is rebooting.

System - Rebooting...

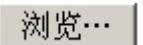
Changes applied.



Waiting for router...

4) Flash new firmware image

Keep settings:

When upgrade a new firmware, check the option Keep settings: , then all the configurations on the PBX will be retained, if user does not check this option, when APX5008 finishes upgrading, all the configurations will be the default ones. And click  to select the firmware user need to upgrade and click  Flash image... , then APX5008 system will ask you to verify the performance, click  Proceed to confirm the action and click  Cancel to cancel.

Flash Firmware - Verify

The flash image was uploaded. Below is the checksum and file size listed, compare them with the original file to ensure data integrity.
Click "Proceed" below to start the flash procedure.

- Checksum: 79577edbe9f6664e66b76afcc3b757028
- Size: 14.44 MB (15.69 MB available)
- Configuration files will be kept.

 Cancel  Proceed

Click Proceed, APX5008 will finish upgrading automatically and reboot itself. It takes several minutes for APX5008 to finish the upgrading, please be patient.

Notice: do not power off when APX5008 is upgrading, or the upgrading may fail or the file may be damaged.

2.2.2.9 Reboot

Click **System--> Reboot** to login the reboot page of APX5008, click [Perform reboot](#) , then APX5008 will reboot.

System

Reboot

Reboots the operating system of your device

[Perform reboot](#)

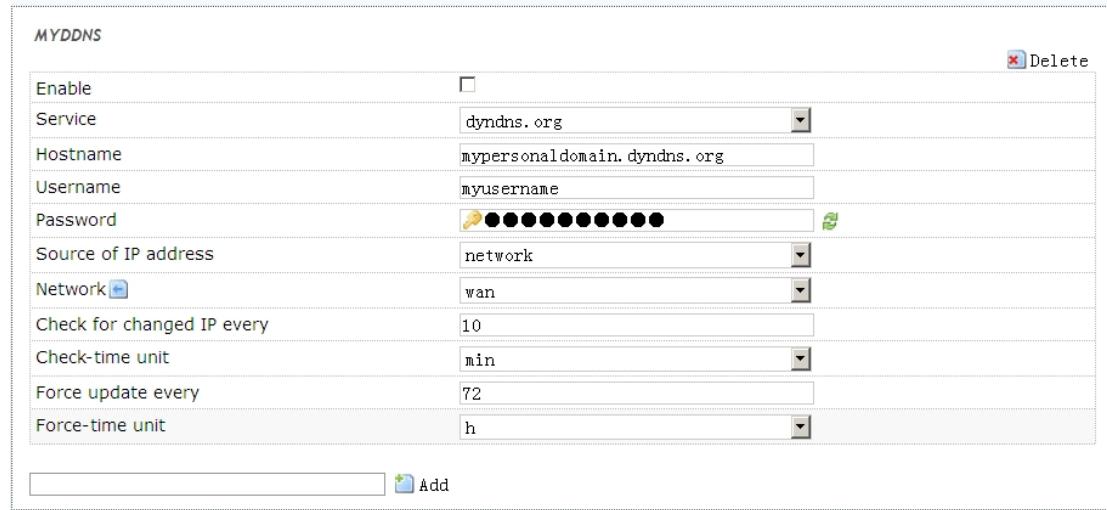
2.2.3 Services

2.2.3.1 Dynamic DNS

Dynamic DNS allows that your router can be reached with a fixed hostname while having a dynamically changing IP address. Configurations are as following:

Dynamic DNS

Dynamic DNS allows that your router can be reached with a fixed hostname while having a dynamically changing IP address.



| MYDDNS | |
|----------------------------|-----------------------------|
| Enable | <input type="checkbox"/> |
| Service | dyndns.org |
| Hostname | mypersonaldomain.dyndns.org |
| Username | myusername |
| Password | ████████████████ |
| Source of IP address | network |
| Network | wan |
| Check for changed IP every | 10 |
| Check-time unit | min |
| Force update every | 72 |
| Force-time unit | h |

 Add

- ◆ Enable: check this option, the DDNS you configured can take effect, or the DDNS won't take effect.
- ◆ Service: Enter the service provider which supports DDNS service to you.
- ◆ Hostname: Enter the domain name that you get from the provider.
- ◆ Username: Enter the username that you get from the provider.
- ◆ Password: Enter the password that you get from the provider.
- ◆ Source of IP address: Configure the source of IP address which you would like to bind the domain name to, such as Interface, URL and network.
- ◆ Check for changed IP every: set a time interval for the system to check if the IP changed, every time interval, the device will check whether the IP of the interface which binds to the DDNS changes, if it does, the system will send the new IP address to the server.
- ◆ Check-time unit: set the unit for the checking changed IP interval, minute or hour, for example.
- ◆ Force update every: In the setting time, the APX5008 will make force update on the IP and send the IP to the DDNS server.
- ◆ Force-time unit: set the unit for force update time, minute or hour, for example.

Additionally, APX5008 system support multiple DDNS, user can click  Add to add several DDNS.

2.2.4 Network

2.2.4.1 DHCP/DNS

Click **Network--> DHCP/DNS** to login the configuration page. In this page, user can configure some parameters of DHCP and DNS, including NAT and firewall.

DHCP and DNS

Dnsmasq is a combined DHCP-Server and DNS-Forwarder for NAT firewalls

Server Settings

| General Settings | Resolv and Hosts Files | TFTP Settings | Advanced Settings |
|-------------------|--|---------------|-------------------|
| Domain required | <input checked="" type="checkbox"/> Don't forward DNS-Requests without DNS-Name | | |
| Authoritative | <input checked="" type="checkbox"/> This is the only DHCP in the local network | | |
| Local server | /lan/ Local domain specification. Names matching this domain are never forwarded and resolved from DHCP or hosts files only | | |
| Local domain | lan Local domain suffix appended to DHCP names and hosts file entries | | |
| Log queries | <input type="checkbox"/> Write received DNS requests to syslog | | |
| DNS forwardings | /example.org/10.1.2.3 List of DNS servers to forward requests to | | |
| Rebind protection | <input checked="" type="checkbox"/> Discard upstream RFC1918 responses | | |
| Allow localhost | <input checked="" type="checkbox"/> Allow upstream responses in the 127.0.0.0/8 range, e.g. for RBL services | | |
| Domain whitelist | ihost.netflix.com List of domains to allow RFC1918 responses for | | |

Active Leases

| Hostname | IPv4-Address | MAC-Address | Leasetime remaining |
|----------|--------------|-------------|---------------------|
| | | | |

- ◆ General Settings: Configure the parameters of DNS.
- ◆ Resolv and Hosts Files: Configure the parameters of DHCP.
- ◆ TFTP Settings: The tftp server is embedded in the APX5008 system, this can be used to be network TFTP to share the resources.
- ◆ Advanced Settings: In this page, there are some advanced settings on DHCP and DNS, such as DNS query port, Max. DHCP leases, Max. EDNS0 packet size and so on.

2.2.4.2 Hostnames

Click **System--> Hostnames** to login the hostname configuration webpage, where you can set the host name to the client. Please enter the hostname user named in the hostname option and enter the IP address of the device which is connected to the APX5008, click Save&Apply.

Hostnames

Host entries

| Hostname | IP address |
|----------|------------------------------------|
| PC | 192.168.30.115 (6c:f0:49:c0:73:ad) |

Click Add to add multiple hostnames.

2.2.4.3 Static Routes

User can set the data forwarding routing manually in Static Routes, which is suitable for the simple and relatively stable network environment. Click **Network-->Static Routes** to login the configuration page of Static Routes.

Routes

Routes specify over which interface and gateway a certain host or network can be reached.

| Static IPv4 Routes | | | | | |
|--------------------|--|-----------------|--------------|--------|------|
| Interface | Target | IPv4-Netmask | IPv4-Gateway | Metric | MTU |
| lan | Host-IP or Network if target is a network | 255.255.255.255 | | 0 | 1500 |

- ◆ Interface: Select the interface which is used to forwarding the data.
- ◆ Target: Enter the target of data forwarding, which can be an single IP address or an IP network, if it is a network, please enter the netmask of the network.
- ◆ IPv4-Netmask: If the target is a network, please enter the netmask of the network.
- ◆ IPv4-Gateway: If the data can not be transmitted directly, please enter the gateway address to forward the data to the packet.
- ◆ Metric: Please enter how many routings the data should pass through before arriving in the gateway.
- ◆ MTU: Enter the packet length limit of the forwarding data.

The parameters of IPv6 are all the same with IPv4 except that user does not need to enter the subnet mask.

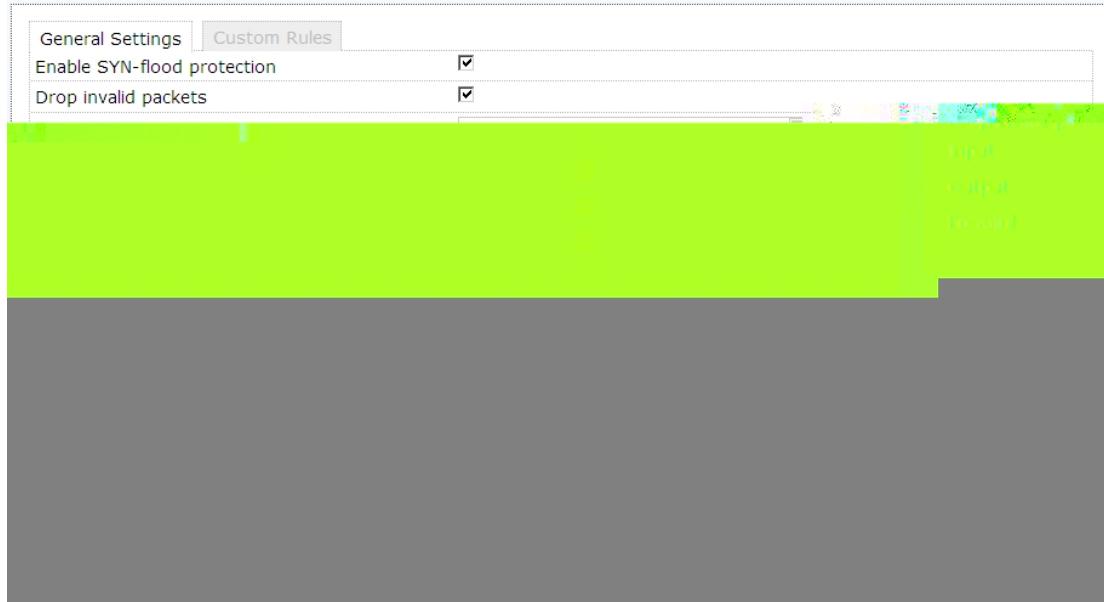
2.2.4.4 Firewall

Firewall settings are very important, these relates to not only the data access among the internal network, but also the data transmission between the internal network and the outside network, and play an important role in network security.

Click **Network--> Firewall** to login the configuration webpage of APX5008's firewall.

Firewall

The firewall creates zones over your network interfaces to control network traffic flow.



There are four parts: General Settings, Zones, Redirections and Rules.

General Settings

- ◆ Enable SYN-flood protection: if or not enable SYN-flood;
- ◆ Drop invalid packets: if or not drop invalid packets;
- ◆ Input: The handling way to the input data, including accept, drop and reject.
- ◆ Output: The handling way to the output data, including accept, drop and reject.
- ◆ Forward: The handling way to the forwarding data, including accept, drop and reject.

Zones

— Zones

| Zone → Forwardings | Input | Output | Forward | Masquerading | MSS clamping | |
|---------------------|--------|--------|---------|-------------------------------------|-------------------------------------|--|
| lan: lan: = wan | accept | accept | accept | <input type="checkbox"/> | <input type="checkbox"/> | |
| wan: wan: = ACCEPT | accept | accept | accept | <input checked="" type="checkbox"/> | <input checked="" type="checkbox"/> | |

- ◆ lan=>wan Input: The handling way to the input data from LAN port, including accept, drop and reject.
- ◆ lan=>wan Output: The handling way to the output data from LAN port, including accept, drop and reject.

- ◆ lan=>wan Forward: The handling way to the forwarding data from LAN port, including accept, drop and reject.
- ◆ wan=>ACCEPT Input: The handling way to the input data from WAN port, including accept, drop and reject.
- ◆ wan=>ACCEPT Output: The handling way to the output data from WAN port, including accept, drop and reject.
- ◆ wan=>ACCEPT Forward: The handling way to the forwarding data from WAN port, including accept, drop and reject.

Redirections

| Redirections | | | | | | |
|---|----------|--------|-----|-------------|--------|------|
| Name | Protocol | Source | Via | Destination | Action | Sort |
| <i>This section contains no values yet</i> | | | | | | |
|  Add | | | | | | |

Click Add to login the configuration page of redirections, including port forwarding and DMZ and so on.

Traffic Redirection

Traffic redirection allows you to change the destination address of forwarded packets.

| General Settings | | Advanced Settings |
|--------------------------|---|-------------------|
| Name | apx5008-01 | |
| Source zone | <input type="radio"/> lan: lan:  <input checked="" type="radio"/> wan: wan:  | |
| Protocol | TCP+UDP | |
| External port | <input type="checkbox"/> Match incoming traffic directed at the given destination port or port range on this host | |
| Internal IP address | <input type="checkbox"/> Redirect matched incoming traffic to the specified internal host | |
| Internal port (optional) | <input type="checkbox"/> Redirect matched incoming traffic to the given port on the internal host | |

- ◆ Name: Enter the name of redirections you are configuring.
- ◆ Source Zone: Configure the accept port of the data redirections.
- ◆ Protocol: Configure the protocol of data redirections.
- ◆ External Port: Matching incoming traffic directed at the given destination port or port range on this host.
- ◆ Internal IP address: Redirect matched incoming traffic to the specified internal host.
- ◆ Internal port(optional): Redirect matched incoming traffic to the given port to the internal port.

Rules

Click “Add” to login the corresponding webpage to add new rules, where user can configure the details of the firewall rules.

Advanced Rules

Advanced rules let you customize the firewall to your needs. Only new connections will be matched. Packets belonging to already open connections are automatically allowed to pass the firewall.



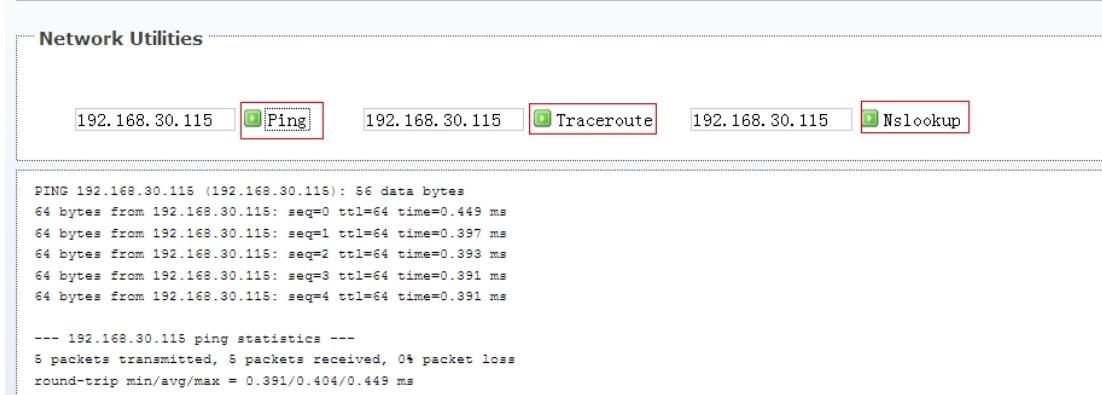
- ◆ Name(Optional): Enter the name of the rules, it can be left blank.
- ◆ Source Zone: Select the port which the new rule is applied to.
- ◆ Protocol: Select the protocols which apply to the rules.
- ◆ Source address: Enter the source address which the rule is applied to.
- ◆ Destination address: Enter the destination address which the rule is applied to.
- ◆ Action: Select the action to inform when the traffic is in line with the rule.

2.2.4.5 Diagnostics

APX5008 provides user with three commonly used network diagnostic tool: Ping, Traceroute and Nslookup.

Click **Network--> Diagnostic** to login the diagnostic interface, input the test IP or domain name into the text box, click on the back button, then the diagnostic results will be displayed soon.

Diagnostics



```

PING 192.168.30.115 (192.168.30.115) 56 data bytes
64 bytes from 192.168.30.115: seq=0 ttl=64 time=0.449 ms
64 bytes from 192.168.30.115: seq=1 ttl=64 time=0.397 ms
64 bytes from 192.168.30.115: seq=2 ttl=64 time=0.393 ms
64 bytes from 192.168.30.115: seq=3 ttl=64 time=0.391 ms
64 bytes from 192.168.30.115: seq=4 ttl=64 time=0.391 ms

--- 192.168.30.115 ping statistics ---
5 packets transmitted, 5 packets received, 0% packet loss
round-trip min/avg/max = 0.391/0.404/0.449 ms
    
```

2.2.4.6 QoS

Rank the traffic data packet according to the network IP address, port or service.

Click **Network--> QoS** to login the configuration page of QoS.

Quality of Service

With QoS you can prioritize network traffic selected by addresses, ports or services.

Interfaces

WAN

| | | |
|-------------------------|--------------------------|---|
| Enable | <input type="checkbox"/> |  |
| Classification group | default |  |
| Calculate overhead | <input type="checkbox"/> | |
| Half-duplex | <input type="checkbox"/> | |
| Download speed (kbit/s) | 1024 | |
| Upload speed (kbit/s) | 128 | |

 Add

Classification Rules

| Target | Source host | Destination host | Service | Protocol | Ports | Number of bytes | Sort |
|--------|-------------|------------------|---------|----------|------------------------|-----------------|---|
| priori | all | all | all | all | 22, 53 | |    |
| normal | all | all | all | TCP | 20, 21, 25, 80, 110, 4 | |    |
| expres | all | all | all | all | 5190 | |    |
| priori | all | all | rtp | all | all | |    |
| priori | all | all | sip | all | all | |    |

 Add

Interface

- ◆ Enable: Check the option to enable the speed limit of APX5008's WAN port.
- ◆ Classification group: The configuration group which the network interface limits.
- ◆ Calculate overhead: Calculate the proportion of the upload and download, in order to prevent the saturation of the link.
- ◆ Half-duplex: Check the option to make the wan port work in half-duplex mode.
- ◆ Download speed: Limit the download speed of the WAN port.
- ◆ Upload speed: Limit the upload speed of the WAN port.

Input the the name of the port user need to configure and click Add to edit the port's classification group, speed and so on.

LAN

| | | |
|-------------------------|--------------------------|---|
| Enable | <input type="checkbox"/> |  |
| Classification group | default |  |
| Calculate overhead | <input type="checkbox"/> | |
| Half-duplex | <input type="checkbox"/> | |
| Download speed (kbit/s) | | |
| Upload speed (kbit/s) | | |

Click Delete to remove the current configuration of the interface.

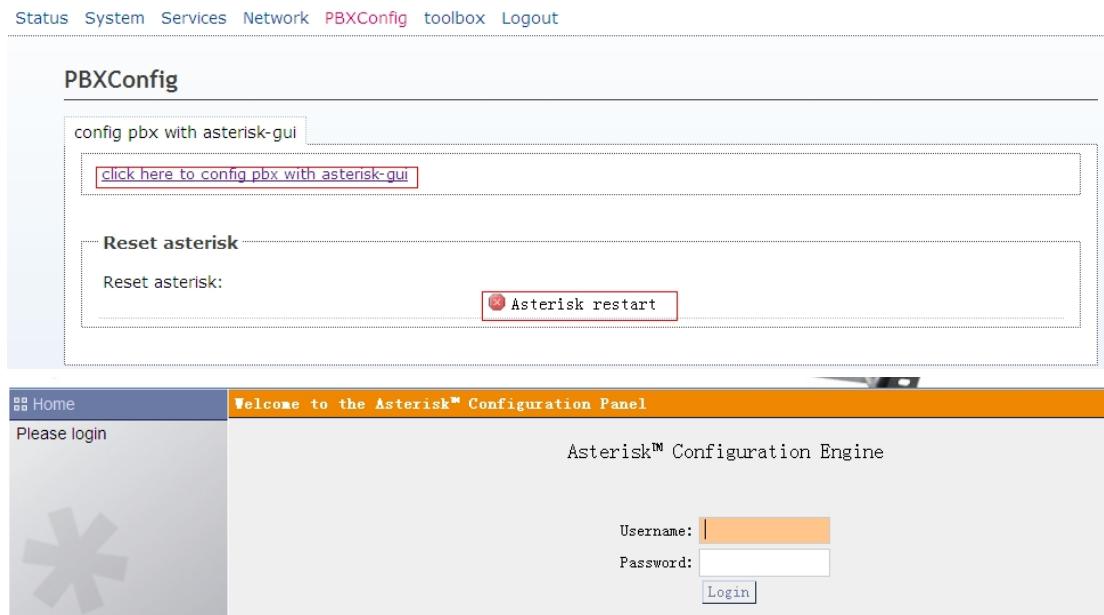
Classification Rules

Classification rules can categorize the priority of all the data, data with high priority should be transmitted in priority, there are 4 levels, priority, express, normal and low. Parameters are as following:

- ◆ Target: Set the priority of the rules.
- ◆ Source host: Select the IP address of the source host, then user can limit the speed of the customized IP.
- ◆ Destination host: Select the IP address of the destination host, then user can limit the speed of the customized IP.
- ◆ Service: Select the services.
- ◆ Protocol: Specify the protocol, including TCP, UDP, ICMP and all.
- ◆ Ports: Specify the ports
- ◆ Number of bytes: Specify how many bites the rule limits, and there is no limit when the traffic overloads the amount.
- ◆ Sort: You can change the order of the rules, rules on the top will be performed firstly.

2.2.5 PBX Config

Click PBXConfig to login the PBX configuration page of APX5008. In this page, press [click here to config pbx with asterisk-gui](#) to login the configuration page, and press **Asterisk restart** to reboot asterisk.



2.2.6 Toolbox

APX5008 provides a series of upload tools on this page, including Asterisk Music On Hold file upload tool, Asterisk back up file upload tool, Asterisk Voice menu prompts file upload tool and asterisk data file upload.

1) Music on Hold file upload

This is used to upload the Hold music file to the APX5008, and the file format should be wav or gsm.

Asterisk Music On Hold file upload

Upload a wav/gsm file to asterisk (music on hold) directory.

| | |
|-----------------|---|
| moh class name: | general |
| Image: | <input type="text" value="C:\Documents and Settings\"/> 浏览... <input type="button" value="upload file..."/> |

Input the music on hold class in **moh class name** option and then press **浏览...** to select the music file from your local PC and then press **upload file...** to upload the file. User can login the PBX page Music on Hold column to check if the file has been uploaded successfully.

2) Back up file upload

This function is used to upload the backup file to the APX5008 system, and user can login the PBX page Backup column to check if the file has been uploaded successfully or not .

Asterisk back up file upload

Upload asterisk backup file.

| | |
|--------|--|
| Image: | <input type="text"/> 浏览... <input type="button" value="upload file..."/> |
|--------|--|

Press **浏览...** to select the back up file and press **upload file...** to upload the file.

3) Voice menu prompts file upload

This function is used to upload the voice menu file to APX5008 system, and user can login the PBX page to check if the file upload successfully or not in Voice Menus column. And the file format should be wav or gsm.

Asterisk Voice menu prompts file upload

Upload a wav/gsm file to asterisk (Voice menu prompts) directory.

| | |
|--------|--|
| Image: | <input type="text"/> 浏览... <input type="button" value="upload file..."/> |
|--------|--|

Press **浏览...** to select the voice menu file and press **upload file...** to upload the file.

4) Asterisk data file upload

This tool is used to upload the voice packets to APX5008 system, and the file format should be

tar.gz. Press **浏览...** to select the file and press **upload file...** to upload the file.

asterisk data file upload

Upload a (tar.gz) file to sdcard.

| | |
|--------|--|
| Image: | <input type="text"/> 浏览... <input type="button" value="upload file..."/> |
|--------|--|

Notice: Please press Save&Apply every time you finish configuring, in order to make the configurations effective, or you can press Save to save the changes and then press Save&Apply to make all the configurations take effect.

3 IP PBX

3.1 Configure the device via GUI

3.1.1 Access the GUI

The default LAN IP address for APX508 is 192.168.0.1, put 192.168.0.1:8088 in your web browser and it will redirect to the PBX configuration page of APX5008, the default username and password for the web access is:

Username: admin

Password: admin

Notice: If you can't access the APX5008, please check whether you have connected the RJ45 cable to the LAN port and whether your computer is in the same network 192.168.0.XXX as the APX5008.

3.1.2 System Status

When you have entered the APX5008 configuration page, the system status will be showed and you can see the system status as below:



| Trunks | | | | |
|---------|-----------------------|--------|----------|-----------------------|
| Status | Trunk | Type | Username | Port/Hostname/IP |
| Request | FlyingVoice | sip | 8888 | 192.168.0.200 |
| Request | ToLilysen | sip | 9999 | 192.168.0.202 |
| | Ports 1,2,3,4,5,6,7,8 | Analog | | Ports 1,2,3,4,5,6,7,8 |

| Extensions | | | |
|------------|------------|----------------|--------------|
| Extension | Name/Label | Status | Type |
| 6000 | Spring | Messages : 0/0 | SIP/IAX User |
| 6001 | Aimee | Messages : 0/0 | SIP/IAX User |
| 6002 | Jeff | Messages : 0/0 | SIP/IAX User |
| 6003 | Redcoco | Messages : 0/0 | SIP/IAX User |
| 6004 | Beauty | Messages : 0/0 | SIP/IAX User |

| Conference Rooms | |
|------------------|--------------|
| Caller ID | Channel |
| @ 6150 | - Not In Use |
| @ 6000 | - Not In Use |

| Parking Lot | | | |
|-------------------|---------|-----------|---------|
| Caller ID | Channel | Extension | Timeout |
| No Parked Calls 0 | | | |

| System Info | | | |
|--|--|--|--|
| General Network Memory Disk | | | |
| Hostname: Applicative-IP-PBX | | | |
| OS Version: Linux Applicative-IP-PBX 2.6.39.4 #1493 Fri Mar 22 07:55:45 EDT 2013 mips GNU/Linux | | | |
| Asterisk Build: 2.6.39.4 #1493 Fri Mar 22 07:55:45 EDT 2013 mips GNU/Linux | | | |

3.1.3 Configure Hardware

The Configure Hardware page lists the available telephony ports in your system. You can configure the hardware to comply with your local telephony environment.

Analog Hardware

| Type | Ports | |
|------------------|------------------------|-------------------------------------|
| FXS Ports | -- | |
| FXO Ports | 1, 2, 3, 4, 5, 6, 7, 8 | <input type="button" value="Edit"/> |

Tone Region ⓘ : China

software echo canceller ⓘ : mg2

Reset all Previous Digital Trunks Information

Advanced Settings

Module Name: wctdm24xxp

Opermode ⓘ : CHINA

a-law override ⓘ : ulaw

fxs honor mode ⓘ : apply opermode to fxo modules only

bostringer ⓘ : normal

fastringer ⓘ : normal

lowpower ⓘ : normal

ring detect ⓘ : standard

MWI mode ⓘ : None

|

Notice: Hover on the (i) and you can see the comment of every settings.

3.1.4 Trunks

Trunks are used to make outbound call to the real world. There are different trunks we can set here.

Manage Analog trunks

Analog Trunks **VOIP Trunks**

| Trunk | Analog Ports | |
|-----------------------|-----------------|---|
| Ports 1,2,3,4,5,6,7,8 | 1,2,3,4,5,6,7,8 | <input type="button" value="Edit"/> <input type="button" value="Delete"/> |

There are eight FXO ports on APX5008, user can set eight analog trunks in this setting page, or user can add several ports in a group to set an analog trunk.

Click “Edit” to edit the current analog trunk, click “Delete” to remove the corresponding Trunk.

Click “New Analog Trunk” to add a new analog trunk, user can add only one FXO port or several ports in a trunk; and input the name of the trunk in Trunk Name option, then click “Add” to save the settings, or click “Cancel” to cancel the settings. And the existed trunk will be displayed in

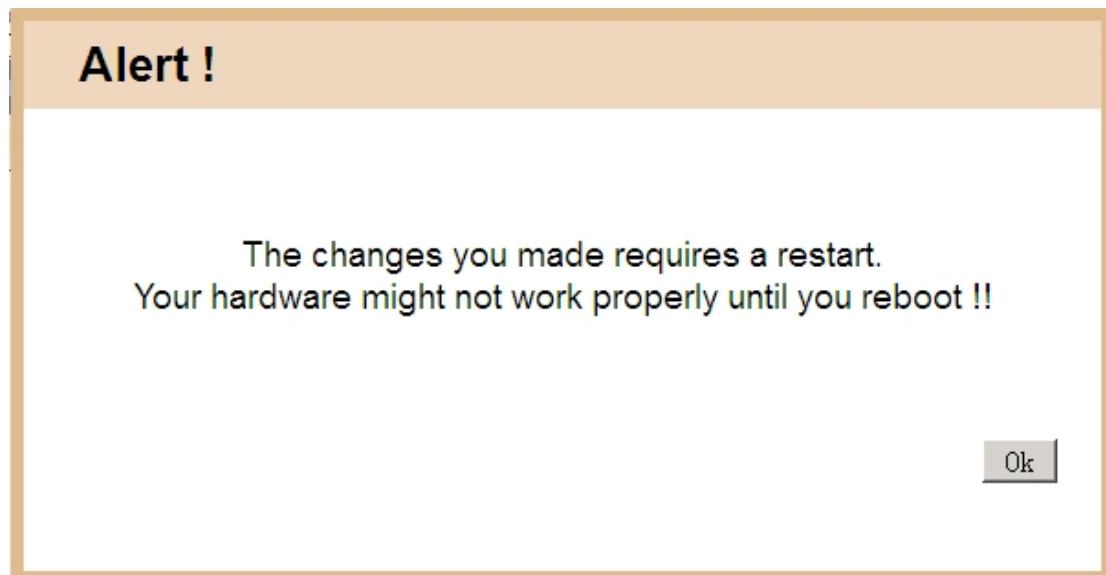
Groups Module.

Manage Analog trunks

New Analog Trunk

| | | | |
|---|---|---|---------------------------------------|
| Channels: | <input type="checkbox"/> 1 <input type="checkbox"/> 2 <input type="checkbox"/> 3 <input type="checkbox"/> 4 <input type="checkbox"/> 5 <input checked="" type="checkbox"/> 6 <input checked="" type="checkbox"/> 7 <input type="checkbox"/> 8 | | |
| Trunk Name ① : | Port6,7 | | |
| Advanced Options | | | |
| Busy Detection ① : | Yes <input type="button" value="▼"/> | Busy Count ① : | 3 |
| Busy Pattern ① : | 500,500 | Ring Timeout ① : | 8000 |
| Answer on ① : | No <input type="button" value="▼"/> | Hangup on ① : | No <input type="button" value="▼"/> |
| Polarity Switch ① : | | Polarity Switch ① : | |
| Call Progress ① : | No <input type="button" value="▼"/> | Progress Zone ① : | US <input type="button" value="▼"/> |
| Use CallerID ① : | Yes <input type="button" value="▼"/> | Caller ID Start ① : | Ring <input type="button" value="▼"/> |
| CallerID ① : | As Received <input type="button" value="▼"/> <input style="width: 100px; height: 20px; border: 1px solid black;" type="text"/> | Pulse Dial ① : | No <input type="button" value="▼"/> |
| CID Signalling ① : | Bell - USA <input type="button" value="▼"/> | mailbox : | <input type="button" value="▼"/> |
| Flash Timing ① : | 750 | Receive Flash Timing ① : | 1250 |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Add"/> | | | |

Click “Apply Changes” to make the trunk effective, and user is required to reboot the APX5008 system.



Please press Ok and reboot APX5008.

VoIP trunks(SIP&IAX2) are also available in the APX5008. Click “New SIP/IAX Trunk” to add a new SIP or IAX trunk in VoIP Trunks page.

Manage SIP & IAX trunks

Analog Trunks **VOIP Trunks**

+ New SIP/IAX Trunk

| Provider Name | Type | Hostname/IP | Username | Edit | Delete |
|---------------|------|---------------|----------|-------------------------------------|---------------------------------------|
| FlyingVoice | SIP | 192.168.0.200 | 8888 | <input type="button" value="Edit"/> | <input type="button" value="Delete"/> |
| ToLilySen | SIP | 192.168.0.202 | 9999 | <input type="button" value="Edit"/> | <input type="button" value="Delete"/> |

Create New SIP/IAX trunk

Type: **SIP**

Context Naming **Assigned by Asterisk GUI**

Provider Name **skype**

Hostname **www.skype.com**

Username **912056320047**

Password **259461xxxx**

Select SIP or IAX in “Type” option, and select the using environment in “Context Naming” option, and fill in the provider name. Then please fill in the Hostname, username and password you get from your SIP /IAX provider. Click “Save” to save the settings or click “Cancel” to cancel your settings.

To make the trunk effective, user needs edit the trunk you just newed, please use the username to fill in the “FromUser” and “AuthUser” options, what is more, please make “insecure” option to be very, and then click “Save” to make the changes effective.

Edit SIP trunk trunk_4

Provider Name **skype**

Hostname **www.skype.com**

Username **912056320047**

Password **259461xxxx**

Codecs: First: **u-law** Second: **a-law** Third: **GSM**
Fourth: **G.726** Fifth:

CallerID

FromDomain:

FromUser: **912056320047**

AuthUser: **912056320047**

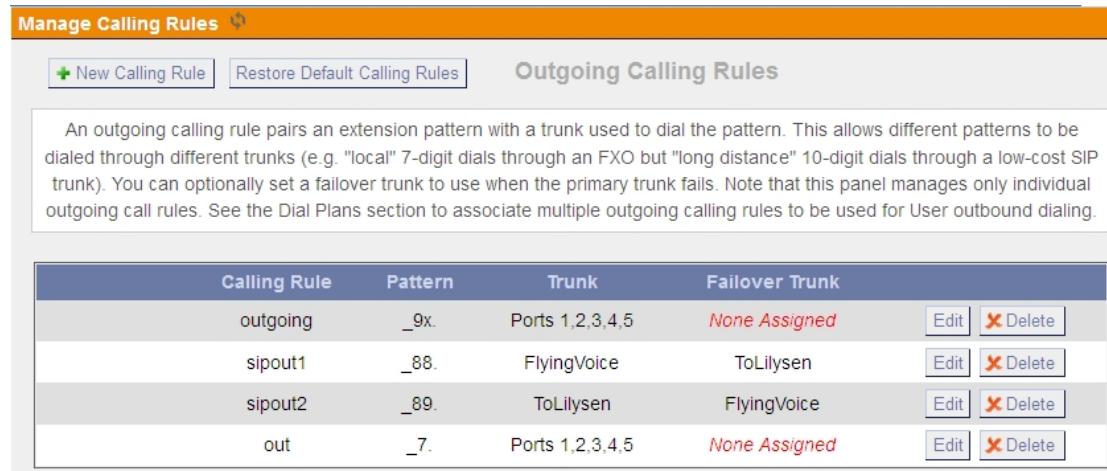
insecure: **very**

Outbound Proxy:

Enable Remote MWI:

3.1.5 Outgoing Calling Rules

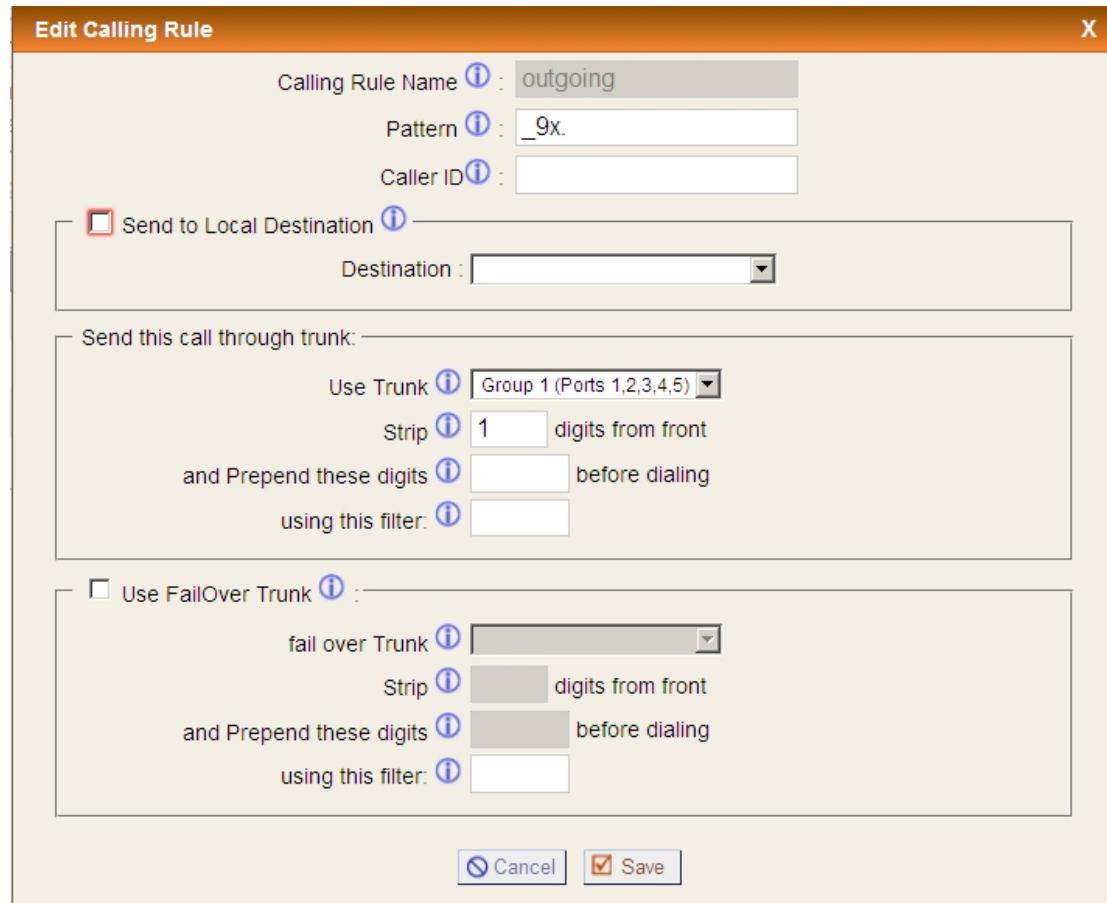
Outgoing Calling Rules define the calling permission and the routing rules when making calls.



The screenshot shows a table titled "Outgoing Calling Rules" with the following data:

| Calling Rule | Pattern | Trunk | Failover Trunk |
|--------------|---------|-----------------|----------------------|
| outgoing | _9x. | Ports 1,2,3,4,5 | <i>None Assigned</i> |
| sipout1 | _88. | FlyingVoice | ToLilysen |
| sipout2 | _89. | ToLilysen | FlyingVoice |
| out | _7. | Ports 1,2,3,4,5 | <i>None Assigned</i> |

Click “New Calling Rule” to add a new calling rule, parameters descriptions are as following:



The "Edit Calling Rule" dialog box contains the following fields and options:

- Calling Rule Name:** outgoing
- Pattern:** _9x.
- Caller ID:** (empty)
- Send to Local Destination:** (unchecked)
 Destination: (dropdown menu)
- Send this call through trunk:**
 - Use Trunk:** Group 1 (Ports 1,2,3,4,5)
 - Strip:** 1 digits from front
 - and Prepend these digits:** (empty) before dialing
 - using this filter:** (empty)
- Use FailOver Trunk:** (unchecked)
 - fail over Trunk:** (dropdown menu)
 - Strip:** (empty) digits from front
 - and Prepend these digits:** (empty) before dialing
 - using this filter:** (empty)
- Buttons:**

- ◆ **Calling Rule Name:** The name of your calling rule
- ◆ **Pattern:** Describe what numbers should use this rule
 - X ... Any Digit from 0-9
 - Z ... Any Digit from 1-9
 - N ... Any Digit from 2-9

[1234-5] ... Any Digit in the brackets(in this example: 1,2,3,4,5,6,7,8,9)

- ... Wildcard, matches anything remaining: i.e. _9011. Matches any number that starts with 9011(excluding 9011 itself)

! ... Wildcard, causes the matching process to complete as soon as it can unambiguously determine that no other matches are possible, for example, the extension _NXXXXXX would match normal 7 digit numbers, while _1NXXNXXXX would represent a three digit area code plus phone number, proceeded by a one.

- ◆ Use/Trunk: Describe which trunk should be used in this rule
- ◆ Strip: Define how many digits should be removed from the dial string.

The following is an example of outgoing calling rule, cut the first digit for all for dial string start with 0, and prepend 86 when dial via this Voipcallout trunk.

Calling Rule Name: Voip_callout

Pattern: _0.

Use Trunk: Voipcallout

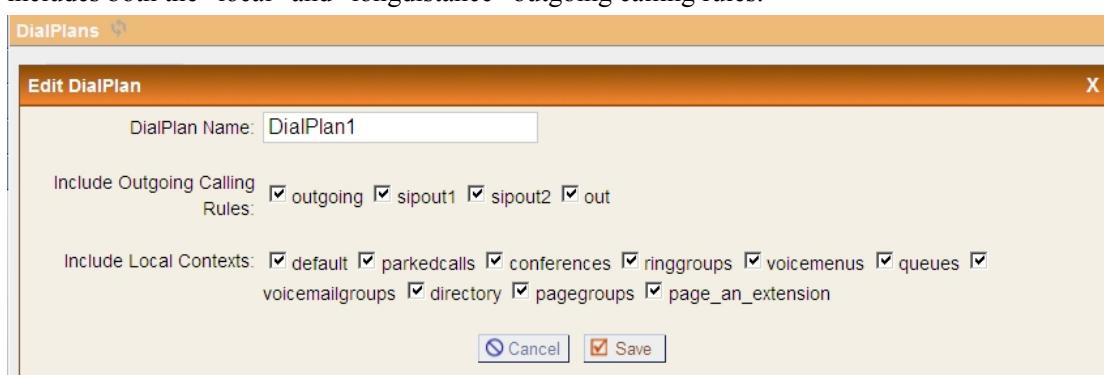
Strip: 1 digit from the front

and prepend these digits: 86

In this case, if you dial 075583545602 using your phone, 8675583545602 will be sent via Voipcallout trunk.

3.1.6 Dial Plan

A Dial Plan is a collection of Outgoing Calling Rules. Dial Plans are assigned to users to specify the dialing permissions they have. For example, you might have one Dial Plan for local calling that only permits users of that Dial Plan to dial local numbers, via the “local” outgoing calling rule. Another user may be permitted to dial long distance numbers, so would have a Dial Plan that includes both the “local” and “longdistance” outgoing calling rules.



3.1.7 Users

User Extensions on PBX

Edit User Extension - 6000

General

Extension: 6000 CallerID Name: Spring DialPlan: DialPlan1
Internal CallerID: 6000 CallerID Number: 6000

Enable Voicemail for this User
VoiceMail Access PIN code: 6000 Email Address:

Technology

SIP IAX Analog Station: None flash: rxflash
Codec Preference: First: u-law Second: a-law Third: GSM Fourth: G.726 Fifth: G.729

VoIP Settings

MAC Address: 6000 Line Number: 1 LineKeys: 1
SIP/IAX Password: 6000 IAX: Require Call Token: auto
IAX: Max Call Numbers:
NAT: Can Reinvite: DTMF Mode: RFC2833 insecure: very

In this page, user can create a list of extensions, user can click “Edit” to modify the settings of the current extension, click “Delete” to remove the corresponding extension, click “Create New User” to add a new extension, select some extensions and click “Modify Selected Users” or “Delete Selected Users” to bulk process the extensions.

Click “Create New User” to add a new extension, parameters are shown in the following picture:

Edit User Extension - 6000

General

Extension: 6000 CallerID Name: Spring DialPlan: DialPlan1
Internal CallerID: 6000 CallerID Number:

Enable Voicemail for this User
VoiceMail Access PIN code: 123456 Email Address:

Technology

SIP IAX Analog Station: None flash: rxflash
Codec Preference: First: u-law Second: a-law Third: GSM Fourth: G.726 Fifth: G.729

VoIP Settings

MAC Address: 6000 Line Number: 1 LineKeys: 1
SIP/IAX Password: xxxxx IAX: Require Call Token: auto
IAX: Max Call Numbers:
NAT: Can Reinvite: DTMF Mode: RFC2833 insecure: very

Other Options

3-Way Calling (analog) In Directory Call Waiting (analog)
 ADA User Is Agent Pickup Group: 1



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General

- ◆ Extension: The number you can dial to reach this user.
- ◆ Name: CallerID name of the user.
- ◆ CallerID: The CID string when you dial to other internal users.
- ◆ Outbound CallerID: Specify the public CallerID for outbound calls, it is only available when your digital or voip provider support this feature.

Voicemail

- ◆ Voicemail Access PIN code: The password for your voicemail box.
- ◆ Email address: The email address for the voicemail to email function.

Technology

- ◆ SIP: Enable this option so the extension can be a SIP device.
- ◆ IAX2: Enable this option so the extension can be an IAX2 device.
- ◆ Analog Station: You can select the port here for your extension if you have analog FXS ports in APX5008.
- ◆ Flash/rxflash: flash parameter for the users.
- ◆ Codec preference: Specify the preference codec for the users.

VoIP Setting

- ◆ MAC address: used for Flyingvoice phone provisioning.
- ◆ Line Number: used for Flyingvoice phone provisioning.
- ◆ Linekeys: used for Flyingvoice phone provisioning.
- ◆ SIP/IAX2 password: user password for SIP/IAX2 registration.
- ◆ NAT: Enable this when you use the APX5008 in public network and the sip devices are in private network.
- ◆ Can Reinvite: Enable this and the APX5008 will try to negotiate the endpoints to router the media string directly(not through APX5008). This can reduce the CPU load of the APX5008 and you will get better voice performance because the media string are sent directly from endpoint to endpoint.
- ◆ DTMF mode: DTMF uses on conversation, the RFC2833 is the most common.

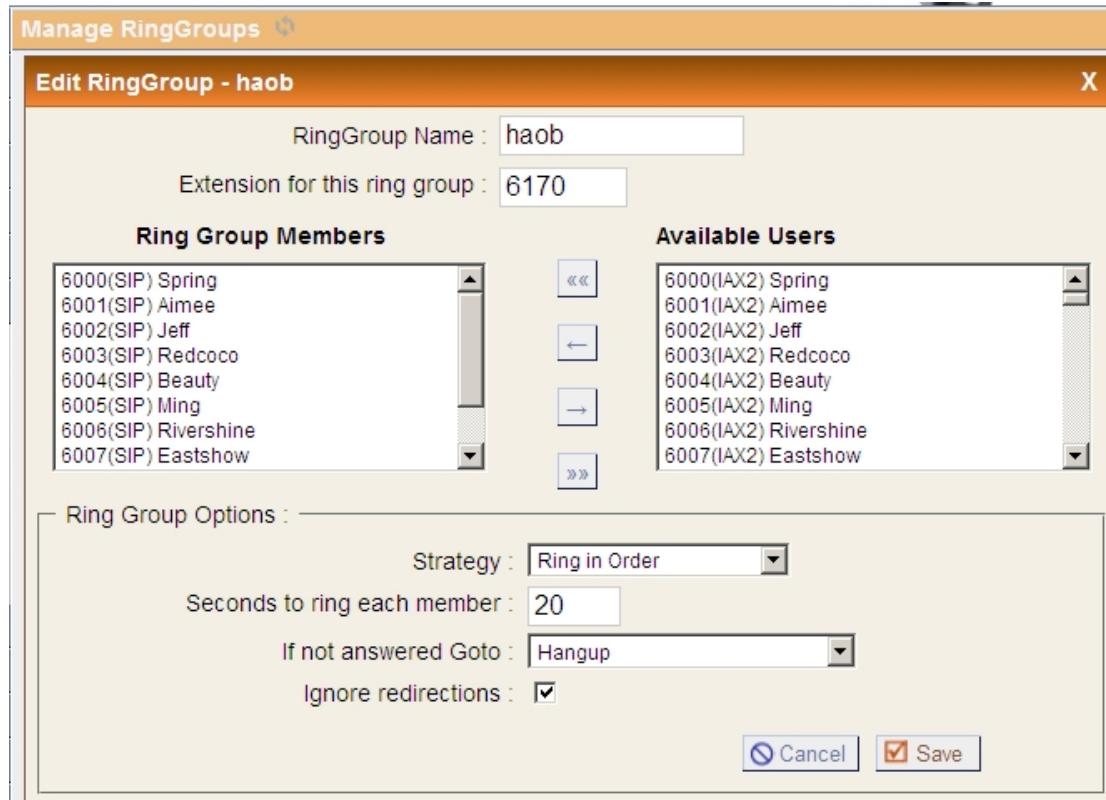
Insecure: method of authentication.

Other Options

- ◆ 3-way Calling: enable/disable 3-way calling.
- ◆ In directory: check this if the user is listed in the directory.
- ◆ Calling waiting: enable/disable call waiting.
- ◆ CTI: Computer Technology Integration, allows access to 3rd party applications over Asterisk Manager Interface.
- ◆ Is Agent: check this if the user is available in call queue.
- ◆ Pick up Group: Specify the call group for the user.

3.1.8 Ring Groups

Define the Ring groups to dial more than one extension simultaneously, or to ring more than one phone sequentially.



Edit RingGroup - haob

RingGroup Name : haob

Extension for this ring group : 6170

| Ring Group Members | | Available Users | |
|----------------------|---|-----------------------|---|
| 6000(SIP) Spring | ↔ | 6000(IAX2) Spring | ↑ |
| 6001(SIP) Aimee | ← | 6001(IAX2) Aimee | ↓ |
| 6002(SIP) Jeff | → | 6002(IAX2) Jeff | ↔ |
| 6003(SIP) Redcoco | ↔ | 6003(IAX2) Redcoco | ↑ |
| 6004(SIP) Beauty | ↔ | 6004(IAX2) Beauty | ↓ |
| 6005(SIP) Ming | ↔ | 6005(IAX2) Ming | ↔ |
| 6006(SIP) Rivershine | ↔ | 6006(IAX2) Rivershine | ↑ |
| 6007(SIP) Eastshow | ↔ | 6007(IAX2) Eastshow | ↓ |

Ring Group Options :

Strategy : Ring in Order

Seconds to ring each member : 20

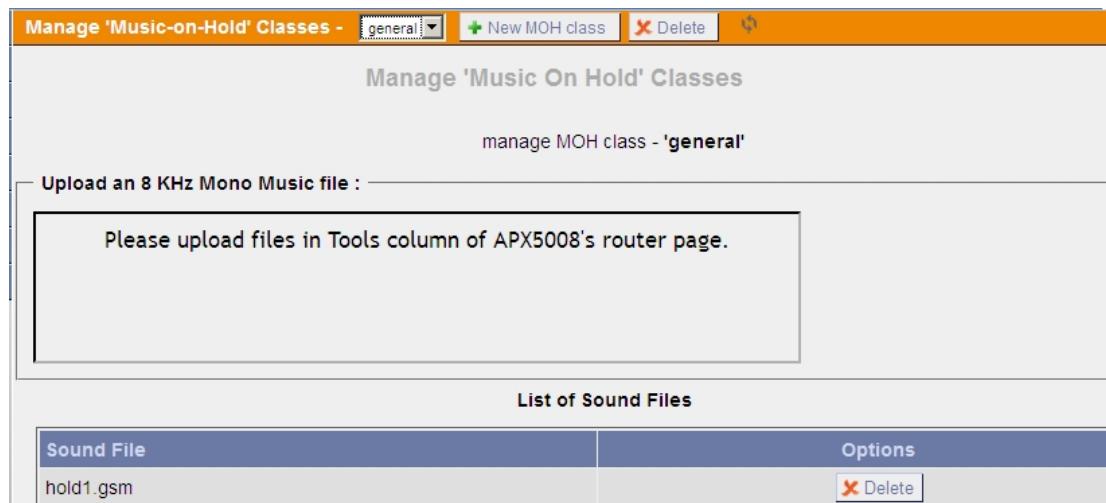
If not answered Goto : Hangup

Ignore redirections :

Name the ring group and enter the extension for the group, and then select the members for the group. Also select the strategy, ringing time, no answer destination and redirections. Click “Save” to save the changes or click “Cancel” to cancel the settings. Please click “Apply Changes” on the upper-right corner to make your changes effective.

3.1.9 Music On Hold

Customize audio tracks for different queues, parked calls etc. To upload music file, please go to APX5008's router page and use the upload tool. User can check the music file in this page.



Manage 'Music-on-Hold' Classes - general

New MOH class

Manage 'Music On Hold' Classes

manage MOH class - 'general'

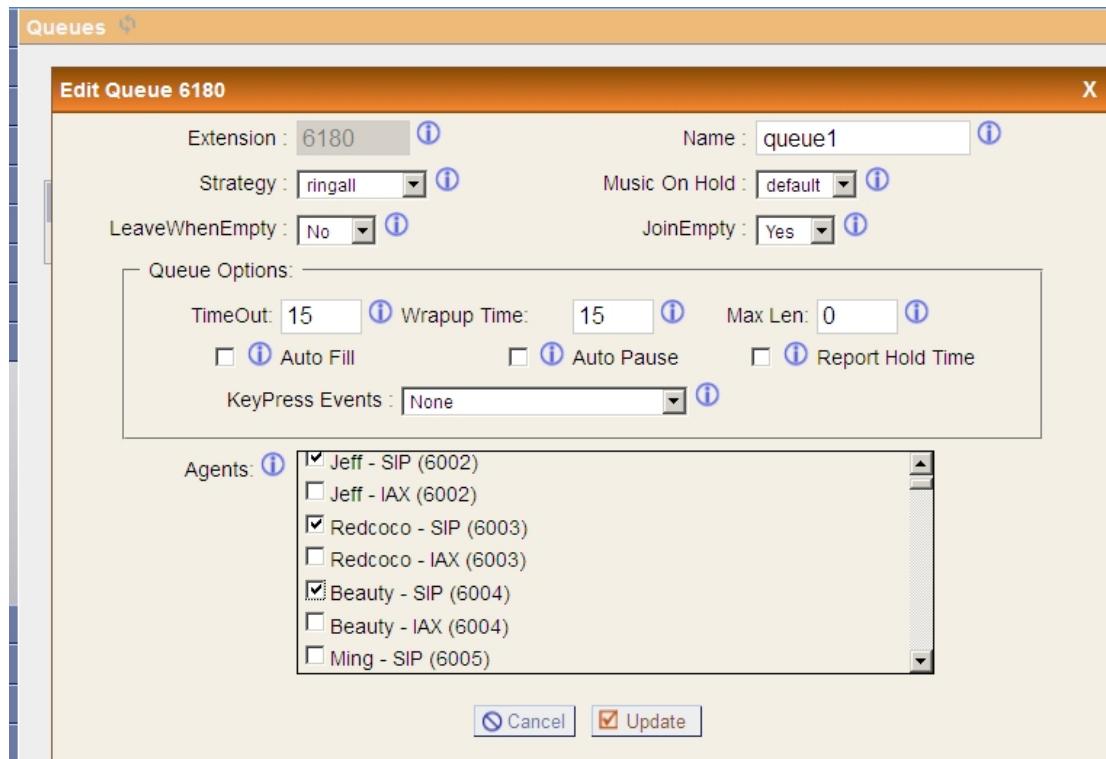
Upload an 8 KHz Mono Music file :

Please upload files in Tools column of APX5008's router page.

| Sound File | Options |
|------------|---------------------------------------|
| hold1.gsm | <input type="button" value="Delete"/> |

3.1.10 Call Queues

Call queues allow calls to be sequenced to one or more agents.



Edit Queue 6180

Extension : 6180 Name : queue1

Strategy : ringall Music On Hold : default

LeaveWhenEmpty : No JoinEmpty : Yes

Queue Options:

TimeOut: 15 Wrapup Time: 15 Max Len: 0
 Auto Fill Auto Pause Report Hold Time
 KeyPress Events : None

Agents:

- Jeff - SIP (6002)
- Jeff - IAX (6002)
- Redcoco - SIP (6003)
- Redcoco - IAX (6003)
- Beauty - SIP (6004)
- Beauty - IAX (6004)
- Ming - SIP (6005)

- ◆ Extension: This option defines the numbered extension that may be dialed to reach this Queue.
- ◆ Name: This option defines a name for this queue, i.e. "Sales". Name is a label to help you see this queue in the queue list.
- ◆ Strategy: This option sets the Ringing Strategy for this Queue. The options are:
 - Ringall: Ring all available agents simultaneously until one answers.
 - RoundRobin: Take turns ringing each available agent.
 - LeastRecent: Ring the agent which was least recently called.



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FewestCalls: Ring the agent with the fewest completed calls.

Random: Ring a random agent.

RRmemory: RoundRobin with Memory, remembers where it left off in the last ring pass.

◆ Agents: This selection shows all users defined as agents in their user conf. Checking a user here makes them a member of the current queue.

◆ Music On Hold: Select the ‘Music on Hold’ class for this queue. ‘Music on Hold’ classes can be managed from the ‘Music On Hold’ panel.

◆ LeaveWhenEmpty: This option controls whether callers already on hold are forced out of a queue which has no agents. There are three options:

Yes: Callers are forced out of a queue when no agents are logged in.

No: Callers will remain in a queue with no agents.

Strict: Callers are forced out of a queue with no agents logged in, or if all logged in agents are unavailable.

◆ Join Empty: This option controls whether callers can join a call queue that has no agents.

There are three options:

Yes: Callers can join a call queue with no agents or only unavailable agents.

No: Callers can not join a queue with no agents.

Strict: Callers can not join a queue with no agents or if all agents are unavailable.

◆ Queue Options:

Timeout: How many seconds an Agent’s phone will ring before the queue tries to ring the next agent.

Wrapup Time: How many seconds after the completion of a call an agent will have before the queue can ring them with a new call. The default is 0, which is no delay.

MaxLen: How many calls can be queued at once. This count does not include calls that have been connected with agents, it only includes calls that have not yet been connected. Default is 0, which is no limit. When the limit has been reached, a caller will hear a busy tone and advance to the next calling rule after attempting to enter the queue.

AutoFill: Defining this option causes the queue, when multiple calls are in it at the same time, to push them to agents simultaneously. Thus, instead of completing one call to an agent at a time, the queue will complete as many calls simultaneously to the available agents.

AutoPause: Enabling this option pauses an agent if they fail to answer a call. This means that the agents is still logged into the queue, but they will not receive calls from the queue. Once paused, an agent can unpause by logging into the queue using the regular agent login extension.

Report Hold Time: Enabling this option causes Asterisk to report, to the agent, the hold time of the caller before the caller is connected to the Agent.

KeyPress Events: If a caller presses a key while waiting in the queue, this setting selects which voice menu should process the key press.

Queues

Agent Login Settings

Agent Login Settings

Agent Login Extension: 900 i

Agent Callback Login Extension: 901 i

To logout of **Agent Login** Hangup your phone. To Logout of **Agent Callback Login** Dial the same extension used to login, specify your extension and password when prompted, and hit # when asked for your callback extension. This will successfully log you out of all queues you are a part of.

Save

- ◆ Agent Login Extension: Extension to be dialed for the agents to login to the specific queue. This is an extension that all the agents can call to login to their specified queues.
- ◆ Agent Callback Login Extension: Extension to be dialed for the agents to login to the queues they are apart of. Same as Agent Login Extension, except you do not have to remain on the line.

3.1.11 Voice Menu(IVR)

Menus allow for more efficient routing of calls from incoming callers. Also known as IVR(Interactive Voice Response) menus or Digital Receptionist.

Manage Voice Menus

Edit VoiceMenu voicemenu-custom-1

Name: **VoiceMenu1** i Advanced Edit

Extension: **700** i

i Allow Dialing Other Extensions

Actions i

| | |
|---|---|
| Answer the call | v ^ x |
| Play /var/lib/asterisk/sounds/record/voicemenu & Listen for KeyPress events | v ^ x |
| Wait '30' sec for the user to enter an extension | v ^ x |

Add new Step: -- Select an Option --

i Allow KeyPress Events

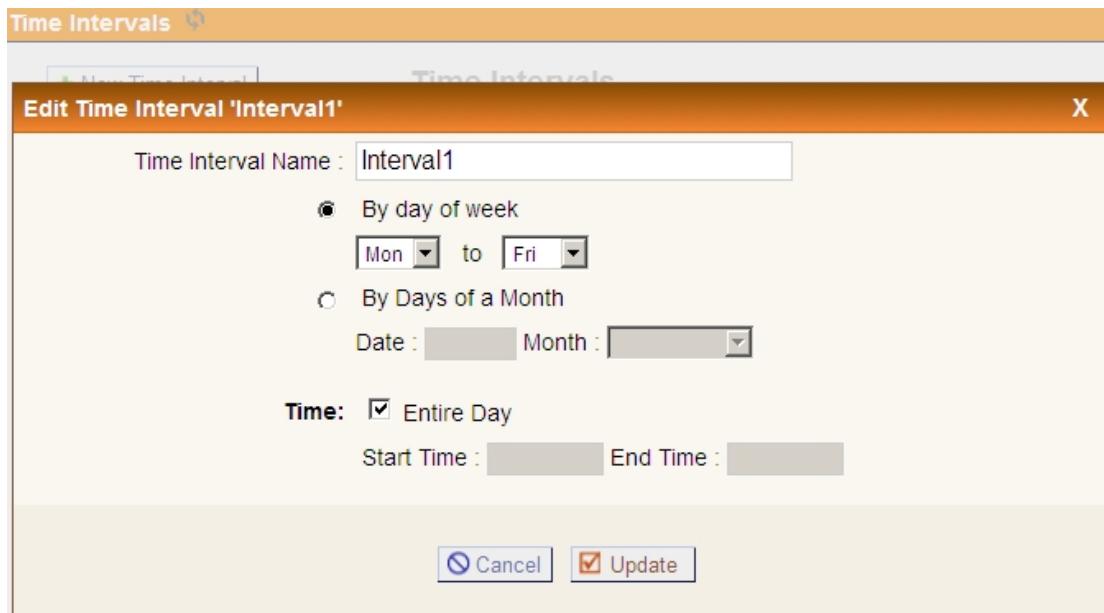
0 Goto User 6000
1 Goto User 6001
2 Goto User 6002
3 Goto User 6003
4 Goto User 6004

- ◆ Name: Name of this voice menu.
- ◆ Extension: If you want this voicemenu to be accessible by dialing an extension, then enter that extension number.
- ◆ Dial other Extensions: Is the caller allowed to dial extensions other than the ones explicitly defined.
- ◆ Actions: A sequence of actions performed when a call enters the menu.

- ◆ Add a new step: Add additional steps performed during the menu.
- ◆ Keypress Events: Allow key press events will cause the system to listen for DTMF input from the caller and defined the actions that occur when a user presses the corresponding digit.

3.1.12 Time Intervals

Time Interval are defined ranges of time that will be used by call routing features.



Edit Time Interval 'Interval1'

Time Interval Name :

By day of week
 By Days of a Month

Mon to

Date : Month :

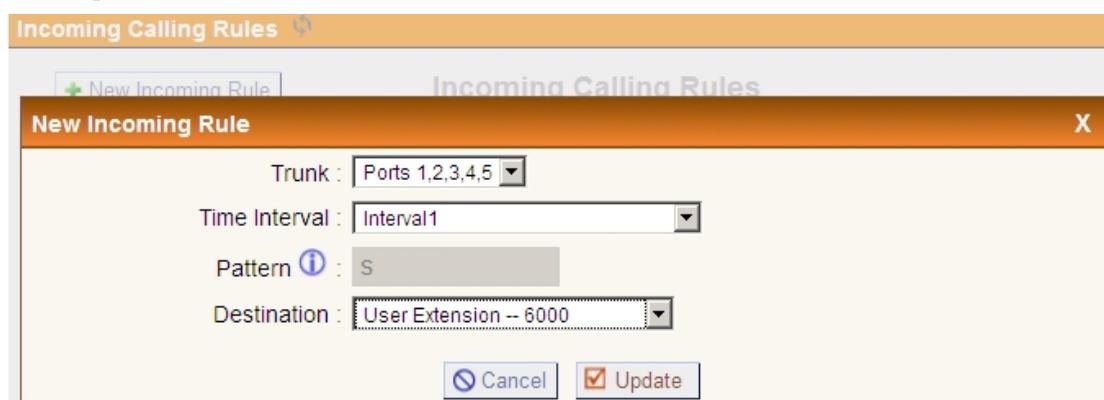
Time: Entire Day

Start Time : End Time :

- ◆ Time Interval Name: Name of the time interval.
- ◆ By day of week: Define the day range by week for time interval.
- ◆ Time/Entire day: Define if the time interval is available for the whole day or only for the specified hours.

3.1.13 Incoming Calling Rules

Create, modify, prioritize and delete incoming call rules for handling incoming calls based on service provider and/or the number called based on time intervals.



New Incoming Rule

Trunk :

Time Interval :

Pattern  : S

Destination :

- ◆ Trunk: Which trunk should be used for the incoming calls.

- ◆ Time Interval: Ranges of time that will be used in this rule,
- ◆ Pattern: Incoming call pattern.
- ◆ Destination: Where the incoming calls should be routed.

3.1.14 Voicemail

General settings for voicemail function.

| | | |
|--|--------------------------------------|----------------------|
| General Settings | Email Settings for VoiceMails | SMTP Settings |
| General VoiceMail Settings | | |
| Extension for checking messages  : <input type="text" value="*97"/> | | |
| Direct Voicemail Dial  : <input type="checkbox"/> | | |
| Max greeting (in seconds)  : <input type="text"/> | | |
| Dial '0' for Operator  : <input type="checkbox"/> | | |
| Message Options | | |
| Maximum messages per folder  : <input type="text" value="10"/> | | |
| Max message time  : <input type="text" value="1 minute"/> | | |
| Min message time  : <input type="text" value="No minimum"/> | | |
| Playback Options | | |
| Say message Caller-ID  : <input type="checkbox"/> | | |
| Say message duration  : <input type="checkbox"/> | | |
| Play envelope  : <input type="checkbox"/> | | |
| Allow users to review  : <input type="checkbox"/> | | |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Save"/> | | |

General Voicemail Settings

- ◆ Extension for checking Message: This option, i.e. “*97” defines the extension that users call in order to access their voicemail accounts.
- ◆ Direct Voicemail Dial: Check this to enable direct voicemail dial. For instance, if Lily’s extension is 6003, you would be able to directly dial into Lily’s voicemail box by dialing #6003 to leave her a message.
- ◆ Max Greeting: Set the maximum number of seconds for a user’s voicemail greeting.
- ◆ Dial’0’ for Operator: Enable callers to exit the voicemail application and connect to an operator extension. The operator extension must be defined on the “Options” panel.



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- ◆ Maximum messages per folder: This select box sets the maximum number of messages that a user may have in any of their folders.
- ◆ Maximum message Time: This select box sets the maximum duration of a voicemail message in seconds. Message recording will not occur for times greater than this amount,
- ◆ Minimum message Time: This select box sets the minimum duration of a voicemail message in seconds. Message below this threshold will be automatically deleted.
- ◆ Say Message Caller-ID: If this option is enabled, the caller ID of the party that left the message will be played back before the voicemail message begins playing.
- ◆ Say Message Duration(in minutes): If this option is set, the duration of the message in mintues will be played back before the voicemail message begins playing.
- ◆ Allow callers to Review: Checking this option allows the caller to review their message before it is submitted as a new voicemail message.
- ◆ Play Envelope: Turn on/off playing introductions about each message when accessing them from the voicemail application.

Email Settings for Voicemails: with this funtion configured, when there is a new voicemail for users, the APX5008 will automatically send the voicemail to the user's email address set in the user's profile.

Voicemail to Email Preference:

- ◆ Send messages by e-mail only: If this option is set, then voicemails will not be checkable using a phone. Messages will be sent via e-mail only.

Notice: You need to have an smtp server configure for this functionality.

- ◆ Attach recording to e-mail: This option defines whether or not voicemails are sent to the user's e-mail as attachments.

Notice: You need to have an smtp server configured for this functionality.

Email Settings for VoiceMails

Send messages by e-mail only [?](#)
 Attach recordings to e-mail [?](#)

Template for Voicemail Emails

From: sales@flyingvoice.com
Subject: New voicemail from \${VM_CALLERID} for \${VM_MAILBOX}
Message: Hello \${VM_NAME}, you received a message lasting \${VM_DUR} at \${VM_DATE} from, (\${VM_CALLERID}). This is message \${VM_MSGNUM} in your voicemail Inbox.

Template Variables: \t : TAB

```

${VM_NAME} : Recipient's firstname and lastname
${VM_DUR} : The duration of the voicemail message
${VM_MAILBOX} : The recipient's extension
${VM_CALLERID} : The caller id of the person who left the message
${VM_MSGNUM} : The message number in your mailbox
${VM_DATE} : The date and time the message was left
    
```

SMTP server setting

SMTP Settings

SMTP设置

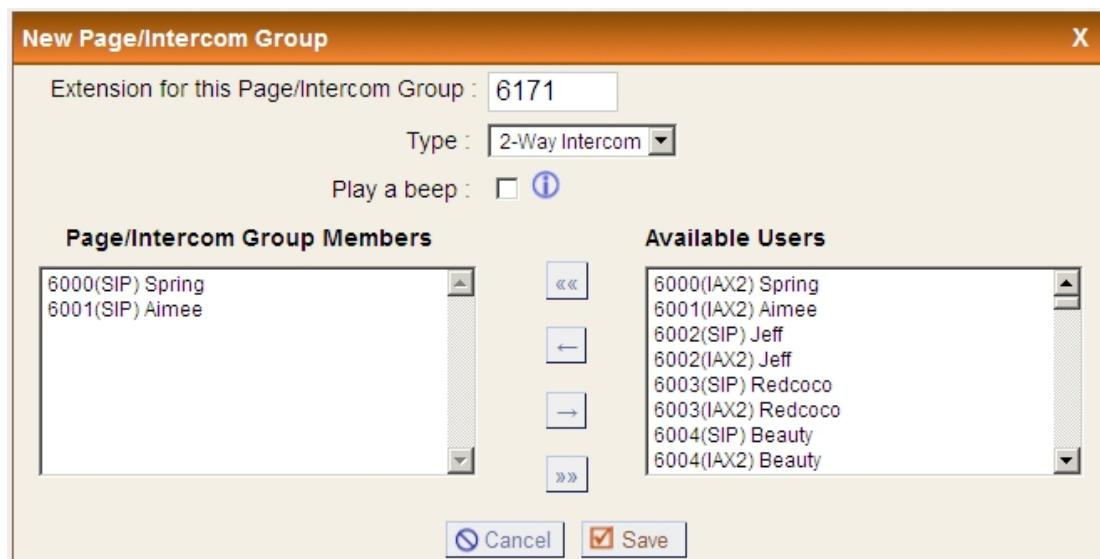
Smtp服务器 [?](#): mail.flyingvoice.com
端口 [?](#):
使用SMTP认证 :
认证用户名 : support
认证密码 : 

- ◆ **SMTP Server:** The IP address or hostname of an SMTP server that your IP PBX may connect to, without authentication, in order to send e-mail notifications of your voicemails, i.e. Mail.yourcompany.com.
- ◆ **Port:** The port number on which the SMTP server is running; generally port 25, if the mail server of your company do not have a port number, leave it blank.

- ◆ Using SMTP Authentication: if or not use SMTP authentication
- ◆ Auth Username: The username of the mailbox on SMTP server, for example support@flyingvoice.com, enter support is OK.
- ◆ Auth Password: Input the password of the mailbox.

3.1.15 Paging/Intercom

A Page/Intercom Group can be used to make an announcement over the speakerphone on a group of phones. Targeted phones will ring one time and answer immediately into speaker-phone mode. Note that this functionality is dependent on a compatible and correctly configured handset. For a user to be able to dial a Page/Intercom group, the ‘pagegroups’ local context must be included in the user’s Dialplan.



- ◆ Extension for the Page/Intercom Group: This is the number dialed to reach this page/intercom group.
- ◆ Type: Select 1-way page or 2-way Intercom.
- ◆ Play a beep: If this option is checked, a beep sound will be played when the intercom call is connected to inform users that they can begin talking.
- ◆ Page/Intercom Groups members: Users which have been added to the Page/Intercom group.
- ◆ Available Users: The users which can be added to the Page/Intercom group.

3.1.16 Conferencing

MeetMe conference bridging allows quick, ad-hoc conferences with or without security.

Manage Conference Rooms

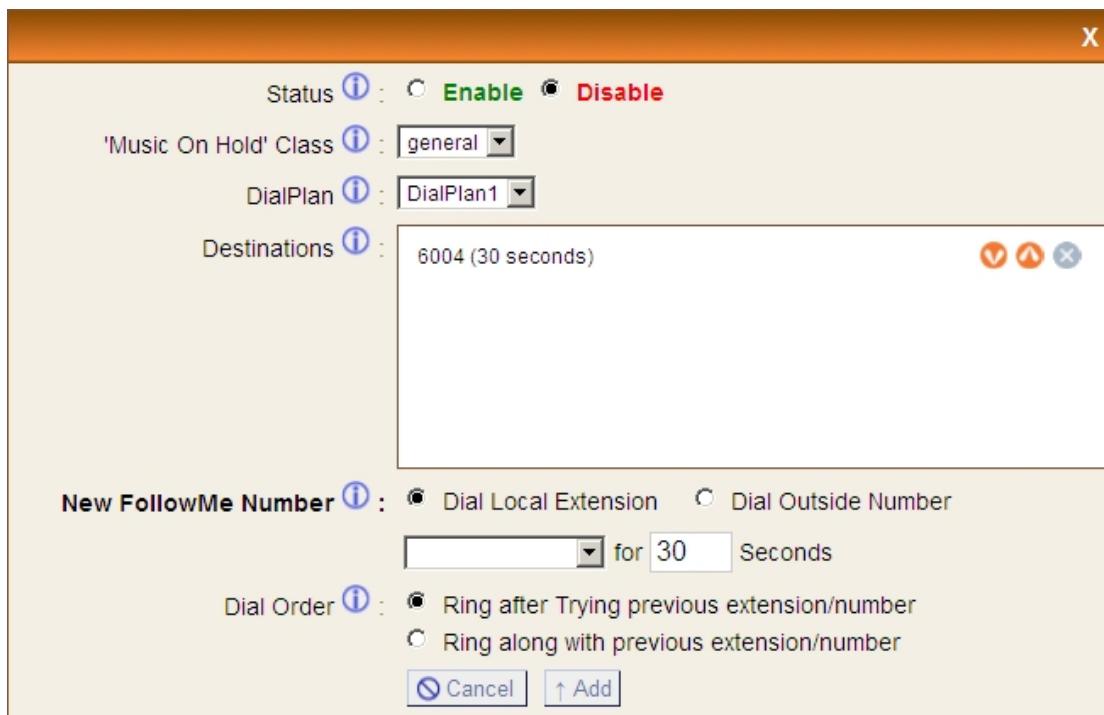
Edit Conference Bridge 6150

| | |
|--|---|
| Extension : <input type="text" value="6150"/> ? | Marked/Admin user Extension : <input type="text" value="6151"/> ? |
| Password Options: | |
| Pin Code: <input type="text" value="123"/> ? | Admin PinCode: <input type="text" value="1234"/> ? |
| Conference Room Options: | |
| <input checked="" type="checkbox"/> ? Play hold music for first caller | <input type="checkbox"/> ? Close conference when last marked user exits |
| <input checked="" type="checkbox"/> ? Enable caller menu | <input checked="" type="checkbox"/> ? Announce callers |
| <input checked="" type="checkbox"/> ? Quiet Mode | <input type="checkbox"/> ? Wait for marked user |
| <input type="checkbox"/> ? Record conference | |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Update"/> | |

- ◆ Extension: This is the number dialed to reach this conference bridge.
- ◆ Marked/Admin user extension: If the conference bridge is to have marked users or admin users, then those users should enter the conference bridge using a separate extension. Admin conference users can lock and unlock the conference and can kick the most recent conference participant. Marked users are special users whose entrance and exit, if the “Wait for Marked user” or “Close conference when the last marked user exits” can either begin or end the conference altogether.
- ◆ Pin Code: set optional pin code, i.e. “1234” that must be entered in order to access the conference bridge.
- ◆ Admin Pin Code: Defining this option sets a PIN for conference administrators.
- ◆ Play hold Music for First Caller: Checking this option causes Asterisk to play hold music to the first user in a conference, until another user has joined the same conference.
- ◆ Enable Caller Menu: Checking this option allows a user to access the conference bridge menu by pressing the * key on their dialpad.
- ◆ Announce Callers: Checking this option announces, to all bridge participants, the joining of any other participants.
- ◆ Quiet Mode: If checked, system won’t play enter/leave sounds.
- ◆ Wait for Marked User: Prevent conference participants from hearing each other until the marked user has joined.

3.1.17 Follow Me

This function enables users to call the other extensions bounded to the custom extension when there is no response in turns or simultaneously.



- ◆ Status: Enable/Disable follow me for this user.
- ◆ Music On Hold class: that the caller would hear while tracking the user.
- ◆ DialPlan: DialPlan that would be used for dialing the FollowMe numbers. By default this would be the same dialplan as that of the user.
- ◆ Destinations: List of extensions/numbers that would be dialed to reach the user during Follow Me.

Click “Add FollowMe Number” to add numbers for this extension.

- ◆ New FollowMe Number: select a local extension or an outside number, input the number in the blank and input the ringing time.
- ◆ Dial Order: Select the dialing order.

3.1.18 Call Features

In this page, user can set call park function, call transfer, call recording function and so on.

Feature Codes & Call Parking Preferences

| Feature Options Feature Digit Timeout: <input type="text" value="1000"/> (milliseconds) | Dial Options <input checked="" type="checkbox"/> t - Allow the called party to transfer the calling party by sending the 'Blind Transfer' or 'Attended Transfer' feature maps. <input checked="" type="checkbox"/> T - Allow the calling party to transfer the called party by sending the 'Blind Transfer' or 'Attended Transfer' feature maps. <input checked="" type="checkbox"/> h - Allow the called party to hang up by sending the 'Disconnect' feature map. <input checked="" type="checkbox"/> H - Allow the calling party to hang up by sending the 'Disconnect' feature map. <input checked="" type="checkbox"/> k - Allow the called party to enable parking of the call by sending the 'Call Parking' feature map. <input checked="" type="checkbox"/> K - Allow the calling party to enable parking of the call by sending the 'Call Parking' feature map. | | | | | |
|--|---|--------|-------------|----------------|-----------|------------------------|
| Call Parking Extension to Dial to Park a Call: <input type="text" value="700"/> Extensions for Parked Calls: <input type="text" value="701-720"/> (Ex: '701-720') Parked Call Timeout (in secs): <input type="text" value="25"/> | Call Options <input checked="" type="checkbox"/> RR - Allow pbx monitor all calls. | | | | | |
| Feature Map Blind Transfer: <input #)<br="" (default="" is="" type="text" value="#*1" }=""/> Disconnect: <input (default="" *)<br="" is="" type="text" value="#*1" }=""/> Attended Transfer: <input <br="" type="text" value="7*"/> Call Parking: <input type="text" value="700"/> | | | | | | |
| Application Map New Application Map | | | | | | |
| Enabled | Feature Name | Digits | ActiveOn/By | App Name | Arguments | |
| <input type="checkbox"/> | Monitor | **6 | Self/Both | Macro | Monitor | Delete |
| <input type="checkbox"/> | stopMonitor | **9 | Self/Both | StopMixMonitor | | Delete |

- ◆ **Parking:** Parking is used to hold a call which has been picked up on extension 700, and then the other extensions registered on the same PBX can dial 701-720 to pick up the call again. If the hold call is not picked up in the limit time, then call can be back on the extension which has parked the call.
- ◆ **Feature Map:** User can set the “Blind Transfer” feature code, “Attended Transfer” feature code , “Disconnect” and “Call Parking” feature code in this column. For example, when user set “Blind Transfer” code to be “#*1”, user can press this code in a call to transfer the call to others.
- ◆ **Dial Options:** If user want to user the features, user need check the corresponding options in this column.
- ◆ **Call Options:** Check this option, all the calls will be recorded on the Asterisk, user can look over the call recording in “Call recording” page. If user does not want to record all the calls but some, user can realize this through “Application Map” without this option checked.
- ◆ **Application Map:** Set this like the upon picture, and enable this option, when in call, user can choose to record the call by pressing the corresponding digits. For example, press “**6” to begin recording and press “**9” or hang up to end the recording.

Notice:

- 1) *If user want to enable the feature, not only should the user input the feature code and corresponding parameters, but also the user should check the corresponding “Dial Options”. By default, all the dial options has been checked. And user can change the feature code.*
- 2) *By default, “Allow pbx monitor all calls” way recording has priority than “Application Map” way recording. If user need not all call recording but some, please disable option “Allow pbx monitor all calls” and enable the “Application Map” with right parameters inputting.*
- 3) *User can check the recordings in “Call Recording” Page.*

3.1.19 Voice Mail Group

Define ‘VoiceMail Groups’ to leave a voicemail message for a grous of users by dialing an extension.



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VoiceMail Groups

New Voice Mail Group

| | | |
|-------------------------------|---|---|
| VoiceMail Group's Extension: | 6190 | |
| Label: | support | |
| User MailBoxes: | <input checked="" type="checkbox"/> 6000 <input type="checkbox"/> 6001 <input type="checkbox"/> 6002 <input type="checkbox"/> 6003 <input type="checkbox"/> 6004 <input type="checkbox"/> 6005 <input type="checkbox"/> 6006 <input type="checkbox"/> 6007 <input checked="" type="checkbox"/> 6008 <input type="checkbox"/> 6009 <input type="checkbox"/> 6010 <input type="checkbox"/> 6011 <input type="checkbox"/> 6012 <input type="checkbox"/> 6013 <input type="checkbox"/> 6014 <input type="checkbox"/> 6015 <input type="checkbox"/> 6016 <input checked="" type="checkbox"/> 6017 <input type="checkbox"/> 6018 <input type="checkbox"/> 6019 <input type="checkbox"/> 6020 <input type="checkbox"/> 6021 <input type="checkbox"/> 6022 <input type="checkbox"/> 6023 <input type="checkbox"/> 6024 <input type="checkbox"/> 6025 <input type="checkbox"/> 6026 <input type="checkbox"/> 6027 <input type="checkbox"/> 6028 <input type="checkbox"/> 6029 <input type="checkbox"/> 6030 <input checked="" type="checkbox"/> 6031 <input type="checkbox"/> 6032 <input type="checkbox"/> 6033 <input type="checkbox"/> 6034 | |
| Extension for VoiceMail Group | Label | Member MailBoxes |
| 6190 | support | 6000, 6008, 6017, 6031 |
| | | Edit Delete |

3.1.20 Voice Menu Prompts

Record or Upload custom VoiceMenu prompts, and user can check the current voice menu prompts on this page.

Custom Voice Menu Prompts

List of Custom Voice Menu Prompts

[Record a new Voice Menu prompt](#) [Upload a Voice Menu prompt](#) [Where to Buy](#)

| # | Name | Options |
|---|---------------|--|
| 1 | voicemenu.gsm | Record Again Play Delete |

Custom Voice Menu Prompts

List of Custom Voice Menu Prompts

[Record a new Voice Menu prompt](#) [Upload a Voice Menu prompt](#)

Record a new Voice Menu prompt

| | |
|--|------------|
| File Name: | voicemenu1 |
| Format: | GSM |
| dial this User Extension to record a new voice prompt: | 6000 |
| Cancel Record | |

Click “Record a new Voice Prompt” to use a phone to record the a piece of voice menu prompt and input the file name, then select the format for the voice file and select the the extension which



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is used to record the prompt, lastly press Record to begin the recording. When finish recording, hang up the phone

User can upload the voice menu prompts in on APX5008's router page, the format of the file must be wav or gsm, and the file will be renamed by the APX5008 system.

3.1.21 Call Recording

User can check the call recording in this page, the recording are separated according to the date.

| List of Recording file | | |
|------------------------|------------|---|
| # | Name | Options |
| 1 | 2013-05-03 | Open Folder>>> |
| 2 | 2013-05-09 | Open Folder>>> |
| 3 | 2013-05-10 | Open Folder>>> |

Press "Open Folder" to view the recording file.

| List of Recording file | | |
|------------------------|------------|---|
| # | Name | Options |
| 1 | 2013-05-03 | Open Folder>>> |
| 2 | 2013-05-09 | Open Folder>>> |
| 3 | 2013-05-10 | Open Folder>>> |

All the file at 2013-05-09

| # | Name | Options |
|---|--------------------------------------|---|
| 1 | 2013-05-09-@-17:11:29--6000-6004.wav | Download Delete |
| 2 | 2013-05-09-@-17:27:45--6000-6004.wav | Download Delete |
| 3 | 2013-05-09-@-17:28:54--6004-.wav | Download Delete |
| 4 | 2013-05-09-@-17:29:10--6004-.wav | Download Delete |
| 5 | 2013-05-09-@-17:30:05--6000-.wav | Download Delete |

And user can download or delete the corresponding recording file.

3.1.22 System Info

User can check the system information there, including general information, network status, disk usage and memory usage.

System Information

- [General](#) [Network](#) [Disk Usage](#) [Memory Usage](#)

OS Version:
Linux Applicative-IP-PBX 2.6.39.4 #1493 Fri Mar 22 07:55:45 EDT 2013 mips GNU/Linux

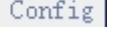
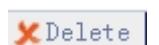
Uptime:
18:26:37 up 14:01,
Load Average: 0.00, 0.01, 0.04

Asterisk Build:
Asterisk/1.8.10.1
Asterisk GUI-version : 2.0

Server Date & TimeZone: Sun Mar 24 18:26:37 CST 2013

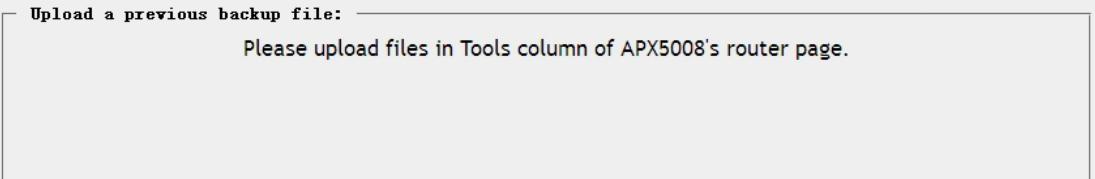
Hostname:
Applicative-IP-PBX

3.1.23 Backup

Backup or restore the previous configuration file. User can upload the backup configuration file in APX5008's router page. Press  to create a new backup, press  to save the backup file to your local computer. Press  to restore the settings and press  to remove the current backup file.

Backup/ Restore Configurations

Manage Configuration Backups

Upload a previous backup file: 

Please upload files in Tools column of APX5008's router page.



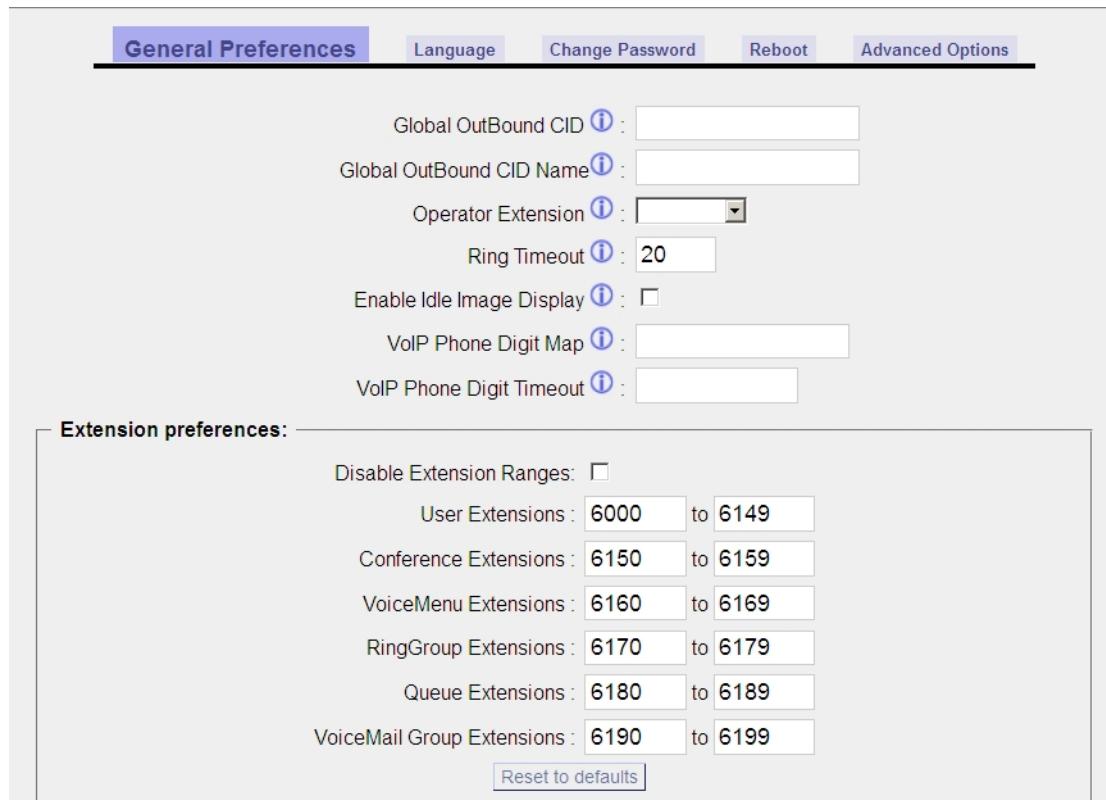
List of Previous Configuration Backups :

| S. No | Name | Date | Options |
|-------|-------------------------|--------------|--|
| 1 | backup_2013apr02_102316 | Apr 02, 2013 |    |

Notice: *Restore the configurations won't take effect immediately on the network setting. You need to modify the network setting in the GUI and save/reboot.*

3.1.24 Options

User can set the extension range, change PBX page language, modify PBX password, Reboot PBX and so on.



General Preferences

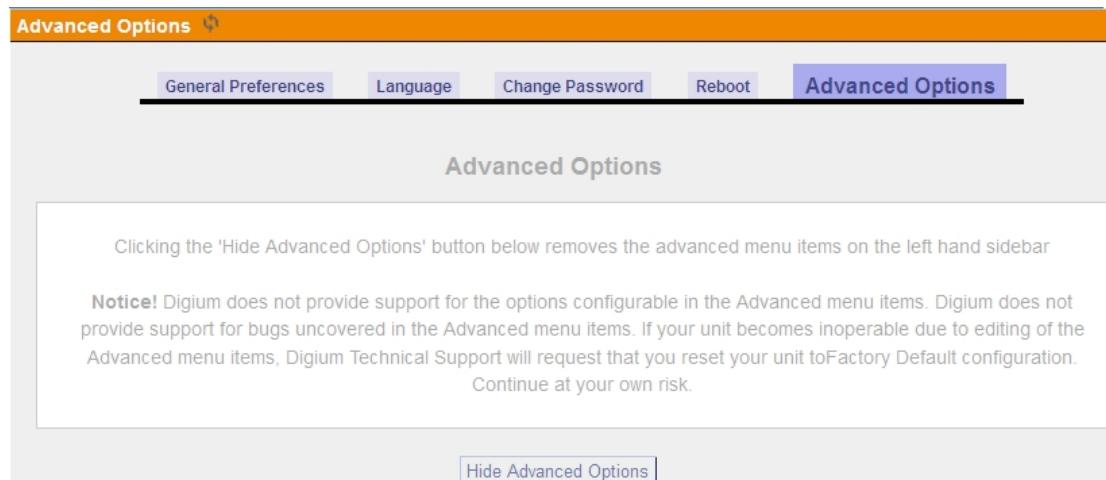
- Global OutBound CID
- Global OutBound CID Name
- Operator Extension
- Ring Timeout 20
- Enable Idle Image Display
- VoIP Phone Digit Map
- VoIP Phone Digit Timeout

Extension preferences:

- Disable Extension Ranges:
- User Extensions: 6000 to 6149
- Conference Extensions: 6150 to 6159
- VoiceMenu Extensions: 6160 to 6169
- RingGroup Extensions: 6170 to 6179
- Queue Extensions: 6180 to 6189
- VoiceMail Group Extensions: 6190 to 6199

3.1.25 Advanced Options

In the options panel, choose Advanced Options-->Show Advanced Options then the advanced options will be showed in the left menu.



Advanced Options

Clicking the 'Hide Advanced Options' button below removes the advanced menu items on the left hand sidebar

Notice! Digium does not provide support for the options configurable in the Advanced menu items. Digium does not provide support for bugs uncovered in the Advanced menu items. If your unit becomes inoperable due to editing of the Advanced menu items, Digium Technical Support will request that you reset your unit to Factory Default configuration.
Continue at your own risk.

The advanced options include:



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Call Detail Records

Active Channels

Bulk Add

File Editor

Asterisk CLI

SIP Settings

3.1.26 Advanced Options -- Call Details Records

Show the call details

| Call Detail Report | | | | | |
|--|----------|--------|-------------|-----------------|-------------|
| <input checked="" type="checkbox"/> Inbound calls ① <input checked="" type="checkbox"/> Outbound calls ① <input checked="" type="checkbox"/> Internal calls ① <input checked="" type="checkbox"/> External calls ① | | | | | |
| <input type="checkbox"/> Show all fields ① <input type="checkbox"/> Show system calls ① | | | | | |
| 4 Total records; Viewing 1-4 of 4 Selected | | | | | |
| Previous Next Click on column header to sort by that column. Click on row to display full record. | | | | | |
| Start time | Duration | Source | Destination | Caller ID | Disposition |
| 1 2013-03-24 11:02:44 | 0:01:48 | 6004 | 6000 | "Beauty" <6004> | ANSWERED |
| 2 2013-03-24 11:02:12 | 0:00:27 | 6004 | 6000 | "Beauty" <6004> | ANSWERED |
| 3 2013-03-24 11:01:44 | 0:00:39 | 6000 | 6004 | "Spring" <6000> | ANSWERED |
| 4 2013-03-24 11:00:58 | 0:01:09 | 6004 | 6000 | "Beauty" <6004> | ANSWERED |

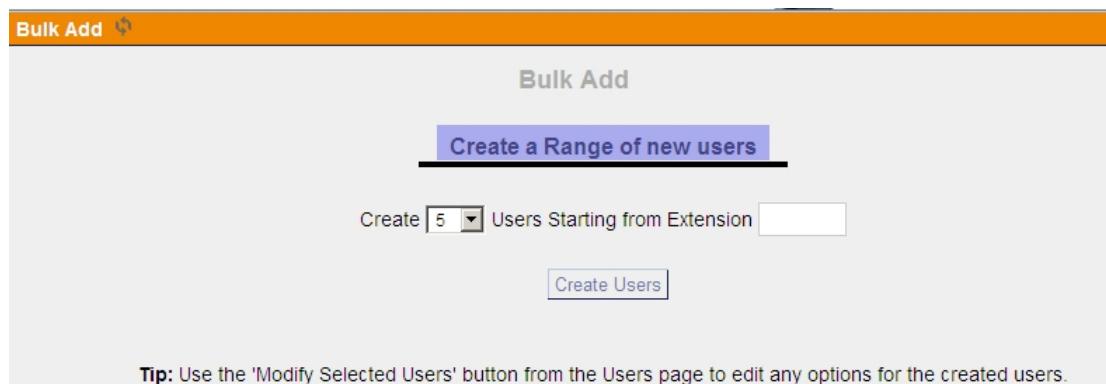
3.1.27 Advanced Options -- Active channels

Show the active channels on the PBX and user can hangup the call or transfer there.

| Active Channels - 4 | | | | | |
|--|-----------|-----------|---|--------------------------|------------------------|
| Refreshing Active Channels in 1 Seconds | | | | | |
| Channel | State | Seconds | Application | Transfer | Hangup |
| SIP/6004-00000006 | undefined | 14 | Dial(\${ARG2}, \${RINGTIME}, \${DIALOPTIONS}) | Transfer | Hangup |
| al/executecommand@asterisk_guitools-f0bb:2 | undefined | 2181 | System(\${command}) | Transfer | Hangup |
| al/executecommand@asterisk_guitools-b246:2 | undefined | 1822 | System(\${command}) | Transfer | Hangup |
| SIP/6000-00000007 | Ringing | undefined | | Transfer | Hangup |

3.1.28 Advanced Options -- Bulk Add

Add a series of users.



Bulk Add

Bulk Add

Create a Range of new users

Create Users Starting from Extension

Tip: Use the 'Modify Selected Users' button from the Users page to edit any options for the created users.

3.1.29 Advanced Options -- File Editor

Here you can modify the asterisk configuration files directly.



File Editor adsi.conf New File

adsi.conf

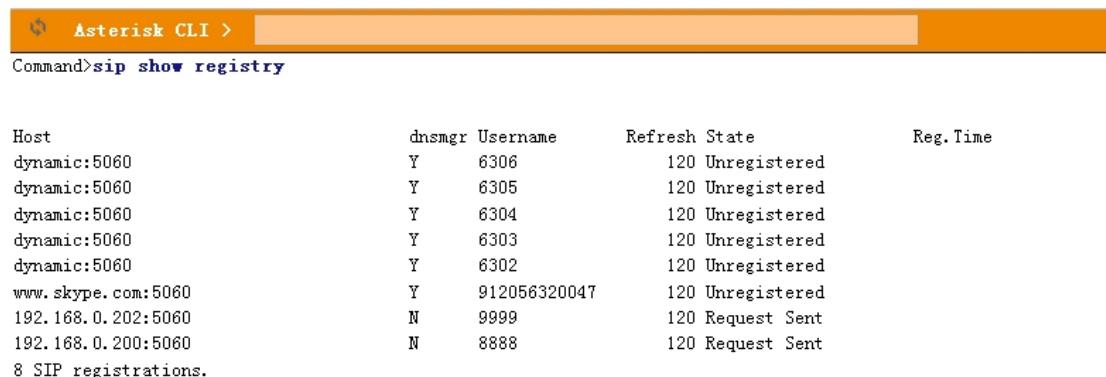
```
- [intro]
alignment=center
greeting=Welcome to the
greeting=Asterisk
greeting=Open Source PBX
```

```
alignment=center
greeting=Welcome to the
greeting=Asterisk
greeting=Open Source PBX
```

Notice: Please make sure you know what the meaning in the files before trying to modify the files.

3.1.30 Advanced Options -- Asterisk CLI

Asterisk CLI interface. Fill the command in the blank and press enter.



Asterisk CLI >

Command>**sip show registry**

| Host | dnsmgr | Username | Refresh | State | Reg. Time |
|----------------------|--------|--------------|---------|--------------|-----------|
| dynamic:5060 | Y | 6306 | 120 | Unregistered | |
| dynamic:5060 | Y | 6305 | 120 | Unregistered | |
| dynamic:5060 | Y | 6304 | 120 | Unregistered | |
| dynamic:5060 | Y | 6303 | 120 | Unregistered | |
| dynamic:5060 | Y | 6302 | 120 | Unregistered | |
| www.skype.com:5060 | Y | 912056320047 | 120 | Unregistered | |
| 192.168.0.202:5060 | N | 9999 | 120 | Request Sent | |
| 192.168.0.200:5060 | N | 8888 | 120 | Request Sent | |
| 8 SIP registrations. | | | | | |

3.1.31 Advanced Options -- SIP settings

Modify the SIP settings of APX5008.

SIP (Session Initiation Protocol) Configuration

- General** **TOS** **DebugNotify** **NAT** **Misc** **Jitter Buffer** **Codecs**

Context: ⓘ

Realm for digest authentication: ⓘ

UDP Port to bind to: ⓘ

IP address to bind to: ⓘ

Domain: ⓘ

Allow guest calls: ⓘ

Overlap dialing support: ⓘ

Allow Transfers: ⓘ

Enable DNS SRV lookups (on outbound calls): ⓘ

Pedantic: ⓘ

SIP Domain Support

From Domain: ⓘ

Auto Domain: ⓘ

Allow External Domains: ⓘ

3.1.32 Asterisk Logs

User can check the asterisk logs here. Click the blank and select the date then press , then all the asterisk logs will be showed.

| Asterisk Log Messages | | | | | | | |
|---|-----|-----|-----|-----|-----|-----------|--|
| 24 Mar 2013 | | | | | | | |
| « March 2013 » | | | | | | | |
| Mon | Tue | Wed | Thu | Fri | Sat | Sun | |
| 25 | 26 | 27 | 28 | 1 | 2 | 3 | |
| 4 | 5 | 6 | 7 | 8 | 9 | 10 | |
| 11 | 12 | 13 | 14 | 15 | 16 | 17 | |
| 18 | 19 | 20 | 21 | 22 | 23 | 24 | |
| 25 | 26 | 27 | 28 | 29 | 30 | 31 | |
| <pre>[Mar 24 00:00:22] NOTICE[1720] chan_ [Mar 24 00:00:22] WARNING[1720] acl_ [Mar 24 00:00:22] WARNING[1720] chan_ [Mar 24 00:00:22] ERROR[1720] chan_s_ [Mar 24 00:00:22] NOTICE[1720] chan_ [Mar 24 00:00:24] ERROR[1720] netsoc_ [Mar 24 00:00:24] WARNING[1720] acl_ [Mar 24 00:00:24] WARNING[1720] chan_ [Mar 24 00:00:24] NOTICE[1720] chan_ [Mar 24 00:00:42] WARNING[1720] acl_ [Mar 24 00:00:42] WARNING[1720] chan_ [Mar 24 00:00:42] ERROR[1720] chan_s_ [Mar 24 00:00:42] NOTICE[1720] chan_sip.c: -- Registration for '9999@192.168.0.202' timed out, trying again (Attempt #592) [Mar 24 00:00:42] WARNING[1720] acl.c: Cannot connect [Mar 24 00:00:42] WARNING[1720] chan_sip.c: sip_xmit of 0x8532e8 (len 350) to 192.168.0.200:5060 returned -2: Network is unreachable [Mar 24 00:00:42] ERROR[1720] chan_sip.c: Serious Network Trouble: __sip_xmit returns error for pkt data [Mar 24 00:00:42] NOTICE[1720] chan_sip.c: -- Registration for '8888@192.168.0.200' timed out, trying again (Attempt #593) [Mar 24 00:00:44] ERROR[1720] netsock2.c: getaddrinfo("www.skype.com", "(null)", ...): Name or service not known [Mar 24 00:00:44] WARNING[1720] acl.c: Unable to lookup 'www.skype.com' [Mar 24 00:00:44] WARNING[1720] chan_sip.c: Probably a DNS error for registration to 912056320047@www.skype.com, trying REGISTER again [Mar 24 00:00:44] NOTICE[1720] chan_sip.c: -- Registration for '912056320047@www.skype.com' timed out, trying again (Attempt #594) [Mar 24 00:01:02] WARNING[1720] acl.c: Cannot connect [Mar 24 00:01:02] WARNING[1720] chan_sip.c: sip_xmit of 0x8532e8 (len 350) to 192.168.0.202:5060 returned -2: Network is unreachable [Mar 24 00:01:02] ERROR[1720] chan_sip.c: Serious Network Trouble: __sip_xmit returns error for pkt data [Mar 24 00:01:02] NOTICE[1720] chan_sip.c: -- Registration for '9999@192.168.0.202' timed out, trying again (Attempt #594) [Mar 24 00:01:02] WARNING[1720] acl.c: Cannot connect [Mar 24 00:01:02] WARNING[1720] chan_sip.c: sip_xmit of 0x8532e8 (len 350) to 192.168.0.200:5060 returned -2: Network is unreachable [Mar 24 00:01:02] ERROR[1720] chan_sip.c: Serious Network Trouble: __sip_xmit returns error for pkt data [Mar 24 00:01:02] NOTICE[1720] chan_sip.c: -- Registration for '8888@192.168.0.200' timed out, trying again (Attempt #594)</pre> | | | | | | | |

4 Application Notes

4.1 Different methods to access the PBX

There are several ways to access the APX5008, different ways has different usage. The web/SSH/telnet accesses are based on network connection, and the console port access is via the console cable which allows you to access the devices on a lower level.

Web access

It is the most common way to access the APX5008. Most settings can be done through the web interface. Simply put the device's IP address in your web browser (better use Firefox), <http://192.168.0.1>, for example, and enter the username and password to access the device. The web access username/password is root/admin. And use admin/admin to enter the PBX panel of APX5008 directly entering <http://192.168.0.1:8088> in the web browser.

SSH access

This is the advanced way to access the device, you can use the Putty software to access the device. In the SSH access, you can access the Linux directly and do more advanced Linux setting and debug. The SSH username/password is root/admin.

Console access

The console access is used mainly for developing purpose or in the case when network is down. Running the Hyper Terminal or Minicom in your computer to connect the APX5008, the settings of the console port should be:

Bit per second to 57600; Data bits: 8

Parity: None

Stop bits: 1

Flow control: None

4.2 Make free internal calls

Making internal calls are the basic requirement for a telephony system. Below are the settings for this usage.

At the beginning, we need to add some extensions to make internal calls. Each extension acts as an internal number. There are three types of extensions we can add: SIP, IAX2 and ZAP.

Before setting up the extensions, we need to go to the **Options --> General Preferences** to set the user extensions range. The default user extensions range is 6000~6149.

Extension preferences:

Disable Extension Ranges:

| | | | |
|------------------------------|------|----|------|
| User Extensions : | 6000 | to | 6149 |
| Conference Extensions : | 6150 | to | 6159 |
| VoiceMenu Extensions : | 6160 | to | 6169 |
| RingGroup Extensions : | 6170 | to | 6179 |
| Queue Extensions : | 6180 | to | 6189 |
| VoiceMail Group Extensions : | 6190 | to | 6199 |

[Reset to defaults](#)

Then go to page **Dial Plan --> Create New Dialplan** to create a default Dial Plan.

DialPlans

Edit DialPlan

DialPlan Name:

Include Outgoing Calling
Rules: outgoing sipout1 sipout2 out

Include Local Contexts: default parkedcalls conferences ringgroups voicemenus queues voicemailgroups directory pagergroups page_an_extension

[Cancel](#) [Save](#)

Then go to page **Users-->Create New User** to create the extensions:6001

Create New User

General :

| | | |
|--|---|--|
| Extension: <input type="text" value="6001"/> | CallerID Name: <input type="text" value="FlyingVoice"/> | DialPlan: <input type="text" value="DialPlan1"/> |
| Internal CallerID: 6001 | CallerID Number: | |

Enable Voicemail for this User

VoiceMail Access PIN code: Email Address:

Technology

| | | | | |
|---|---|---|---|--|
| <input checked="" type="checkbox"/> SIP | <input checked="" type="checkbox"/> IAX | Analog Station: <input type="text" value="None"/> | flash: <input type="text" value="750"/> | rxflash: <input type="text" value="1250"/> |
| Codec Preference : First: <input type="text" value="u-law"/> Second: <input type="text" value="GSM"/> Third: <input type="text" value="None"/> Fourth: <input type="text" value="None"/> Fifth: <input type="text" value="None"/> | | | | |

VoIP Settings

| | | |
|---|---|---|
| MAC Address: <input type="text"/> | Line Number: <input type="text" value="1"/> | LineKeys: <input type="text" value="1"/> |
| SIP/IAX Password: <input type="text" value="████████"/> | IAX: Require Call Token: <input type="text"/> | |
| IAX: Max Call Numbers: <input type="text"/> | | |
| NAT: <input checked="" type="checkbox"/> Can Reinvite: <input type="checkbox"/> | DTMF Mode: <input type="text" value="RFC2833"/> | Insecure: <input type="text" value="no"/> |

Other Options

| | | |
|---|---------------------------------------|--|
| <input type="checkbox"/> 3-Way Calling (analog) | <input type="checkbox"/> In Directory | <input type="checkbox"/> Call Waiting (analog) |
| <input type="checkbox"/> ADA User | <input type="checkbox"/> Is Agent | Pickup Group: <input type="text" value="1"/> |

Fill in the extension numbers in **Extension** and fill in CallerID name, numbers and characters are OK and select Dialplan. Also please set a SIP/IAX password for the extension.

Use the same way to create some other extensions and use IP phones to register the accounts, then user can make internal calls to test.

Notice: please pay attention to the password of your extension.

4.3 Make outbound calls to PSTN and Receive outbound calls

There are many kinds of trunking you can use to make outgoing calls. It includes: Analog FXO trunk, SIP trunk, IAX trunk etc.

4.3.1 Use Analog/FXO trunk

1 Analog Hardware Configuration

In order to make FXO port work, user need to configure the Analog Hardware firstly.

Digital Hardware

No Digital Hardware detected !!

Analog Hardware

| Type | Ports | |
|-----------|------------------------|-------------------------------------|
| FXS Ports | -- | |
| FXO Ports | 1, 2, 3, 4, 5, 6, 7, 8 | <input type="button" value="Edit"/> |

Tone Region (i) :

software echo canceller (i) : oslec

Reset all Previous Digital Trunks Information

Advanced Settings

Module Name:

Opermode (i) : CHINA

a-law override (i) : ulaw

fxs honor mode (i) : apply opermode to fxo modules only

bostringer (i) : normal

fastringer (i) : normal

lowpower (i) : normal

ring detect (i) : standard

MWI mode (i) : None

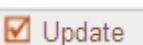
Select “Tone Region” and check “software echo canceller”. Configure the “Opermode” and “a-law override”, what is more, please check the ring detect and configure the value.

2 Create FXO Trunk

Go to page **Trunks-->Add New Analog Trunk**

Edit Analog Trunk

| | | | |
|---|--|---------------------------------------|----------------------------------|
| Channels: | <input checked="" type="checkbox"/> 1 <input checked="" type="checkbox"/> 2 <input checked="" type="checkbox"/> 3 <input checked="" type="checkbox"/> 4 <input checked="" type="checkbox"/> 5 <input type="checkbox"/> 6 <input type="checkbox"/> 7 <input type="checkbox"/> 8 | Groups (i) : | New Group 1 (Ports 1,2,3,4,5) |
| Trunk Name (i) : | Ports 1,2,3,4,5 | | |
| CallerID: | outgoing trunk | | |
| Normally you should not have to adjust your analog ports beyond the initial calibration. Should you still need to fine tune your audio settings, please use the adjustments at the right: | | | |
| Port 1 | Soft | Port 2 | Soft |
| Port 3 | Soft | Port 4 | Soft |
| Port 5 | Soft | | |
| Advanced Options | | | |
| Busy Detection (i) : | Yes | Busy Count (i) : | 3 |
| Busy Pattern (i) : | 350,350 | Ring Timeout (i) : | 8000 |
| Answer on (i) : | No | Hangup on (i) : | No |
| Polarity Switch (i) : | | Polarity Switch (i) : | |
| Call Progress (i) : | No | Progress Zone (i) : | CA |
| Use CallerID (i) : | Yes | Caller ID Start (i) : | Ring |
| CallerID (i) : | As Received | Pulse Dial (i) : | No |

Select the FXO port and fill in the trunk name, click  to add the trunk. Click “Apply Changes” to make the changes effective. And APX5008 system will remind user to reboot Asterisk to make the changes effective, please reboot Asterisk.

Notice: Always click “Apply Changes” in the right top corner when you do some changes.

3 Create Outgoing Calling Rules

New CallingRule

| | |
|--|--|
| Calling Rule Name (i) : | Out_PSTN |
| Pattern (i) : | _9X. |
| Caller ID (i) : | |
| <input type="checkbox"/> Send to Local Destination (i) | |
| Destination : <input type="text"/> | |
| Send this call through trunk: | |
| Use Trunk (i) : | Group 1 (Ports 1,2,3,4,5) <input type="button" value="▼"/> |
| Strip (i) : | 1 digits from front |
| and Prepend these digits (i) <input type="text"/> before dialing | |
| using this filter: (i) <input type="text"/> | |
| <input type="checkbox"/> Use FailOver Trunk (i) | |
| fail over Trunk (i) : | <input type="button" value="▼"/> |
| Strip (i) : | <input type="text"/> digits from front |
| and Prepend these digits (i) <input type="text"/> before dialing | |
| using this filter: (i) <input type="text"/> | |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Save"/> | |

Name the calling rule and fill in the blank, and fill the pattern in the corresponding blank, also select the trunk you need use in “Use Trunk”. If you need failover trunk, you can set it.

The pattern _9X. And strip 1 digit means all calls start with 9 will be cut the first digit and sent out via this rule. For example, if you dial 910668734723, the APX5008 will send 10668734723 to port 1 or port 2 or port 3 or port 4 or port 5.

4 Add New DialPlan

Go to page Dial Plans-->Create New Dial Plans

Create New DialPlan

| | |
|--|--|
| DialPlan Name: | DialPlan2 |
| Include Outgoing Calling Rules: | <input type="checkbox"/> sipout1 <input type="checkbox"/> sipout2 <input type="checkbox"/> out <input checked="" type="checkbox"/> Out_PSTN |
| Include Local Contexts: | <input checked="" type="checkbox"/> default <input checked="" type="checkbox"/> parkedcalls <input checked="" type="checkbox"/> conferences <input checked="" type="checkbox"/> ringgroups <input checked="" type="checkbox"/> voicemenus <input checked="" type="checkbox"/> queues <input checked="" type="checkbox"/> voicemailgroups <input checked="" type="checkbox"/> directory <input checked="" type="checkbox"/> pagergroups <input checked="" type="checkbox"/> page_an_extension |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Save"/> | |

Make the extensions use DialPlan2 and phones register on APX5008 can make outbound call using FXO port 1, port 2, port 3, port 4 and port 5.

5 Create Voice Menu

Go to page Voice Menu-->Create New VoiceMenu.

Manage Voice Menus

Edit VoiceMenu voicemenu-custom-1

Name: IVR 700 Advanced Edit

Extension: 700 Info

Info Allow Dialing Other Extensions

Actions Info

| | |
|---|---|
| Answer the call | V A X |
| Play /var/lib/asterisk/sounds/record/voicemenu & Listen for KeyPress events | V A X |
| Wait '30' sec for the user to enter an extension | V A X |

Add new Step: -- Select an Option --

Info Allow KeyPress Events

- 0 Goto User 6000
- 1 Goto User 6001
- 2 Goto User 6002
- 3 Goto User 6003 Selected
- 4 Goto User 6004

6 Create incoming Calling Rules

Incoming Calling Rules

New Incoming Rule

Trunk: Ports 1,2,3,4,5 ▼

Time Interval: None (no Time Intervals matched) ▼

Pattern Info: S

Destination: VoiceMenu – IVR ▼

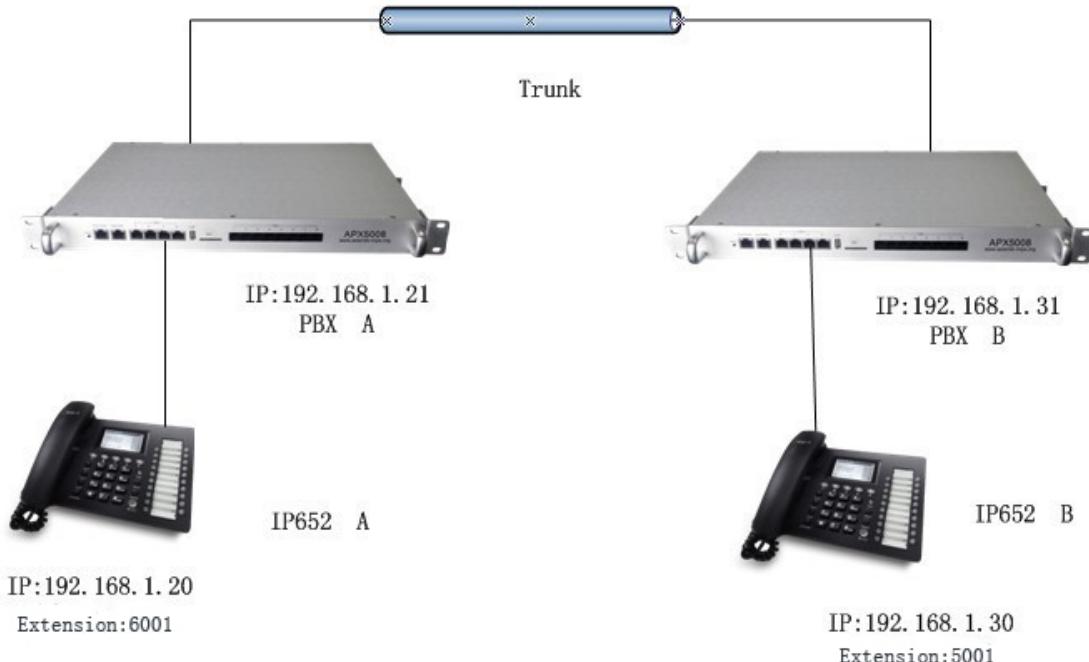
Cancel Update

After these settings, then the outbound phone can make calls to extensions' on APX5008. Please make sure that the PSTN line is connected well to APX5008's FXO port, or calls can not go out, and can not come in either.

4.3.2 Link two PBX in the same network using VoIP Trunk

Connections are as following:

LAN



The chapter is about how to make two IP PBX register VoIP trunk on each other:

- 1) PBX register on PBX B using VoIP trunk, then PBX A can call PBX B using this trunk;
- 2) PBX B register on PBX A using VoIP trunk, then PBX B can call PBX A using this trunk;

In above structure:

- 1) IP652 A registers to PBX A as an extension 6001
- 2) IP652 B registers to PBX B as an extension 5001
- 3) All the extensions under PBX A are in the format 6XXX.
- 4) All the extensions under PBX B are in the format 5XXX.
- 5) Extensions under PBX A can make calls to extension under PBX B use format 5XXX.
- 6) Extensions under PBX B can make calls to extension under PBX A use format 6XXX.
- 7) The two PBX links each other via SIP trunk.

Step 1: Set up a extension 6005 in PBX A

Extension: 6005; phone number of this extension

Name: User_PBXB

Password: 6005; SIP log on password

Caller ID: 6005; Caller ID

Step 2: Set up an SIP trunk in PBX B to link PBX A via this User_PBXB extension. In the page Trunks-->Add Voip Trunk

Create New SIP/IAX trunk

| | |
|--|--------------------------|
| Type: | SIP |
| Context Naming ①: | Assigned by Asterisk GUI |
| Provider Name ①: | To trunk A |
| Hostname ①: | 192.168.1.21 |
| Username ①: | 6005 |
| Password: | 6005 |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Save"/> | |

Step 3: Set Calling Rule in PBX B, all calls start with 6 will be sent to PBX A. In the page: Outgoing Calling Rules-->Add New Calling Rule

New CallingRule

| | |
|--|--|
| Calling Rule Name ①: | To_PBXA |
| Pattern ①: | _6X. |
| Caller ID ①: | |
| <input type="checkbox"/> Send to Local Destination ① | Destination : <input type="button"/> |
| Send this call through trunk: | |
| Use Trunk ①: | To trunk A |
| Strip ① | <input type="button"/> digits from front |
| and Prepend these digits ① | <input type="button"/> before dialing |
| using this filter: ① | <input type="button"/> |
| <input type="checkbox"/> Use FailOver Trunk ① | fail over Trunk ① <input type="button"/> |
| Strip ① | <input type="button"/> digits from front |
| and Prepend these digits ① | <input type="button"/> before dialing |
| using this filter: ① | <input type="button"/> |

Step 4: Add this new calling rule “To_PBXA” to the exist dial plan. In the page: DialPlan-->Edit DialPlan1

Edit DialPlan

| | |
|--|--|
| DialPlan Name: | DialPlan1 |
| Include Outgoing Calling Rules: | <input checked="" type="checkbox"/> sipout1 <input checked="" type="checkbox"/> sipout2 <input checked="" type="checkbox"/> out <input type="checkbox"/> Out_PSTN <input checked="" type="checkbox"/> To_PBXA |
| Include Local Contexts: | <input checked="" type="checkbox"/> default <input checked="" type="checkbox"/> parkedcalls <input checked="" type="checkbox"/> conferences <input checked="" type="checkbox"/> ringgroups <input checked="" type="checkbox"/> voicemenus <input checked="" type="checkbox"/> queues <input checked="" type="checkbox"/> <input checked="" type="checkbox"/> voicemailgroups <input checked="" type="checkbox"/> directory <input checked="" type="checkbox"/> pagegroups <input checked="" type="checkbox"/> page_an_extension |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Save"/> | |

Active the changes and apply the test.

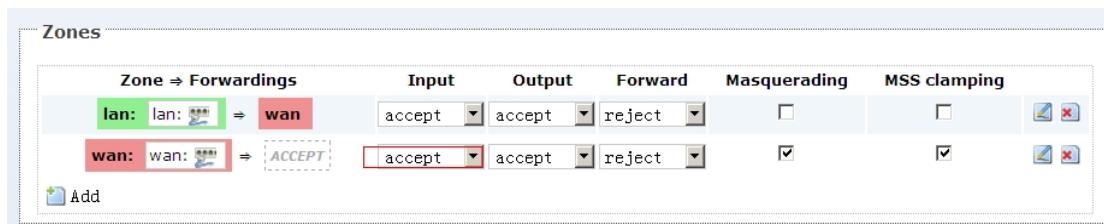
- 1) Register IP652B to PBX B with 5001 extension.
- 2) Register IP652A to PBX A with 6001 extension.
- 3) Use 5001 to dial 6001. And you can see 6001 is ringing and you can pick up the calls.

Above is the way to router PBX B's call to PBX A's call, the method to link PBX A to PBX B is the same as above.

4.3.3 Link two PBX in different location using VoIP trunk

1 Configure the firewall

Login the configuration page of APX5008's Router panel, go to Network-->Firewall page to configure the firewall.



Set "wan=>ACCEPT" column "Input" and "Output" to be accept, then the other PBX can register successfully on this one. Both the PBX need configurations like this.

2 Create extensions

Create an extension on APX5008, the opposite PBX B can use this extension to register, for example, create 6000 on PBX A, PBX B uses 6000 as the SIP trunk account to register on PBX A.

3 Create SIP Trunk

Login the webpage of PBX B, create a SIP trunk use the extension of PBX A, settings are as following:

Edit SIP trunk trunk_5

| | |
|--|---|
| Provider Name ① : | To trunk A |
| Hostname ① : | 192.168.0.230 |
| Username ① : | 6000 |
| Password: | 6000 |
| Codecs: | First : u-law Second : a-law Third : GSM Fourth : G.726 Fifth : |
| CallerID ① : | |
| FromDomain: | |
| FromUser: | 6000 |
| AuthUser: | 6000 |
| insecure: | very |
| Outbound Proxy: | |
| Enable Remote MWI: | <input type="checkbox"/> |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Save"/> | |

4 Create Calling Rule

Login the webpage of PBX B and create new calling rules with the SIP trunk user just created.

Edit Calling Rule

| | |
|--|--------|
| Calling Rule Name ① : | toPBXA |
| Pattern ① : | _6XX. |
| Caller ID ① : | |
| <input type="checkbox"/> Send to Local Destination ① Destination : <input type="button" value="..."/> | |
| Send this call through trunk: Use Trunk ① To trunk A Strip ① 0 digits from front and Prepend these digits ① <input type="button" value="..."/> before dialing using this filter: ① <input type="button" value="..."/> | |
| <input type="checkbox"/> Use FailOver Trunk ① fail over Trunk ① <input type="button" value="..."/> Strip ① <input type="button" value="..."/> digits from front and Prepend these digits ① <input type="button" value="..."/> before dialing using this filter: ① <input type="button" value="..."/> | |
| <input type="button" value="Cancel"/> <input checked="" type="button" value="Save"/> | |

5 Configure DialPlan

Add the calling rules to the DialPlan, make sure the extensions on PBX B use the dialplan.



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Edit DialPlan

DialPlan Name:

Include Outgoing Calling Rules: sipout1 sipout2 out Out_PSTN toPBXA

Include Local Contexts: default parkedcalls conferences ringgroups voicemenus queues voicemailgroups directory pagegroups page_an_extension

6 PBX B configuration

Configure PBX A like PBX Band then user can make calls between PBX A and PBX B to test.